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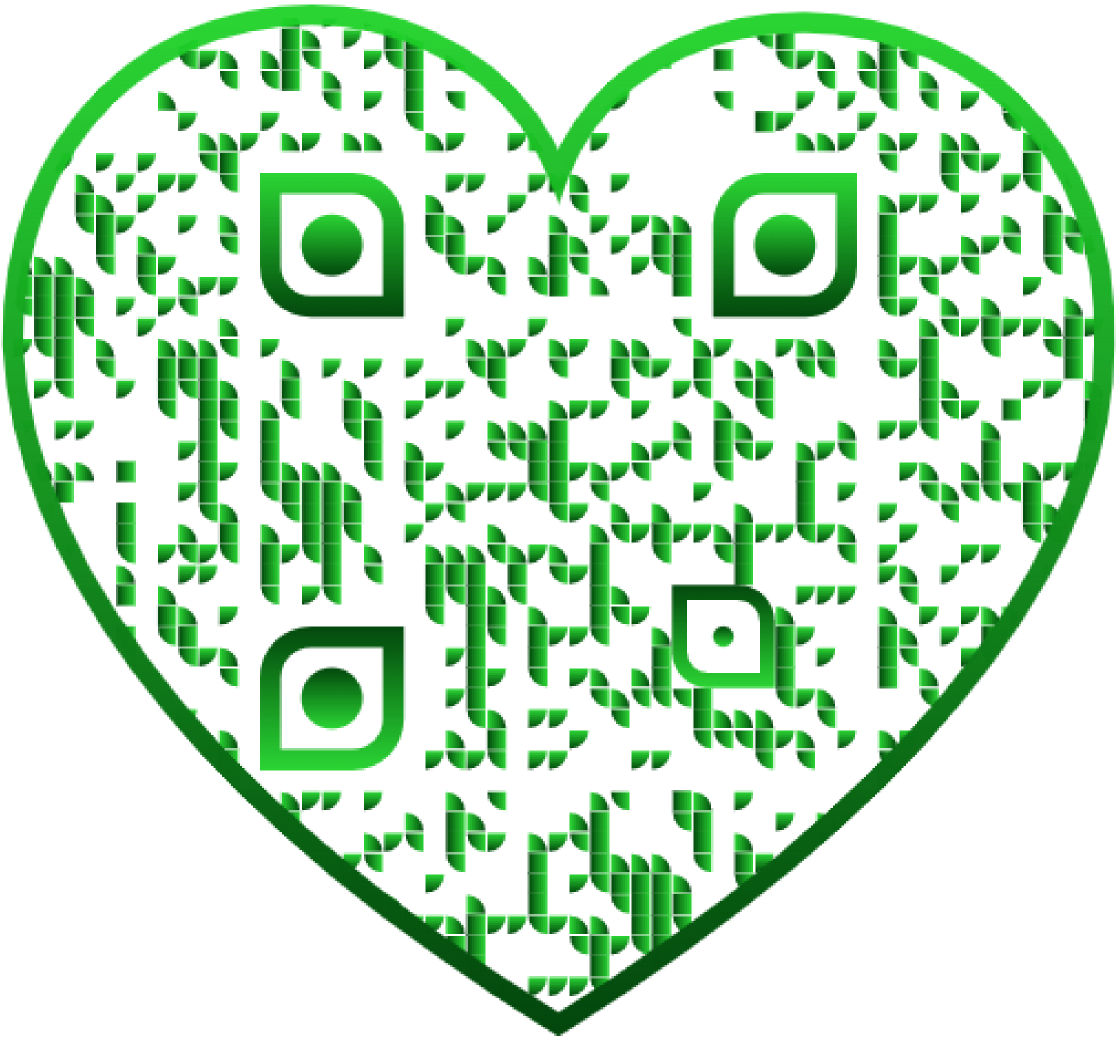

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Audio Industry in 2022

We have wrapped this edition of *audioXpress* with full expectations for CES 2022, hoping to reengage with the people from the many technology companies we regularly write about—including in this edition. However, we don't know at this stage if that will happen. We are sincerely wishing it will and it's important to state that in print for this first edition of 2022. Precisely the one that we are supposed to be handing out at CES.

And given the circumstances we are all going through globally—it is becoming clear that we will continue to be strongly affected by the COVID-19 pandemic throughout 2022, even with all the positive effects of vaccines—I want to use this space to write directly for those companies. About important strategies that I wrote earlier in 2020, and remain particularly relevant two years later.

Stay in Business

Instead of manufacturing tens of thousands of products, sending containers to warehouses, and then generating demand, audio companies should continue to explore the "product drop" concept, which is proving successful given the absence of relevant trade events in the calendar. This is a strategy that focuses on fostering emotional attachment and brand loyalty, but also generates a new type of demand. A brand announces a new innovative and unique product, promotes it on its website, media channels and social media, and schedules a strict deadline for orders. This can be a limited-edition product, but more frequently, by creating this sense of "uniqueness" and urgency the brand is effectively restricting orders to a limited timeframe—after which it ships just the quantities ordered. In a few weeks, months, or a year, the cycle is repeated—unexpectedly, and again with limited availability.

Now, with all of us more receptive to buying online and receiving products at home, this could be even more effective if duly planned. Even for relatively new brands and completely new product categories, the "drop" culture is an effective way to differentiate products in very competitive categories.

Reorganize, focus on product development and other essential activities, and get ready for a full-force return. Communicate more, not less. Work on your product releases and promotion calendars.

Keep your products available. Revisit your supply chain to make sure you can keep up with demand. Expand the online tools and offer the best options. Remember that going online is not the same as direct-to-consumer. You still need to work with a distribution channel to get closer to your clients in every way—for deliveries and technical support. So, make sure your channel partners are also ready.

Make "must have" products a priority and reward your users.

These are strategies that have been validated in the current circumstances and are also reflected in many of the product announcements and market trends that we report in our regular Market Updates.

In this edition, we have tried to look ahead, and show some examples of different areas of innovation in speaker technologies. An interesting take, after we closed this report, was on how the speaker companies are using much more sophisticated engineering approaches through the design and optimization stages. That's almost mandatory to leverage the benefits of some new advanced materials and certainly the use of metamaterials, as illustrated in the examples selected for this edition's report.

Custom-designed and highly engineered approaches to speaker driver designs have been a common practice for most high-volume consumer audio companies, and certainly in automotive audio applications. In recent years, that expanded significantly with companies approaching audio product designs using modeling and simulation tools at the core of the development process, permanently evoked every time experimentation with prototypes show deviations from the intended performance.

The result enables important incremental qualities in products, as long as the correct goals are established earlier at the design stage. The audio industry is increasingly able to tailor speaker performance targets and provide consumers with the best possible sound solutions for a given application.

Maybe the missing link that needs to be addressed is correctly identifying the expectations and preferences of the end users. Clearly, we can create much better audio products than were ever possible, we just need to agree on what really matters.

J. Martins
Editor-in-Chief

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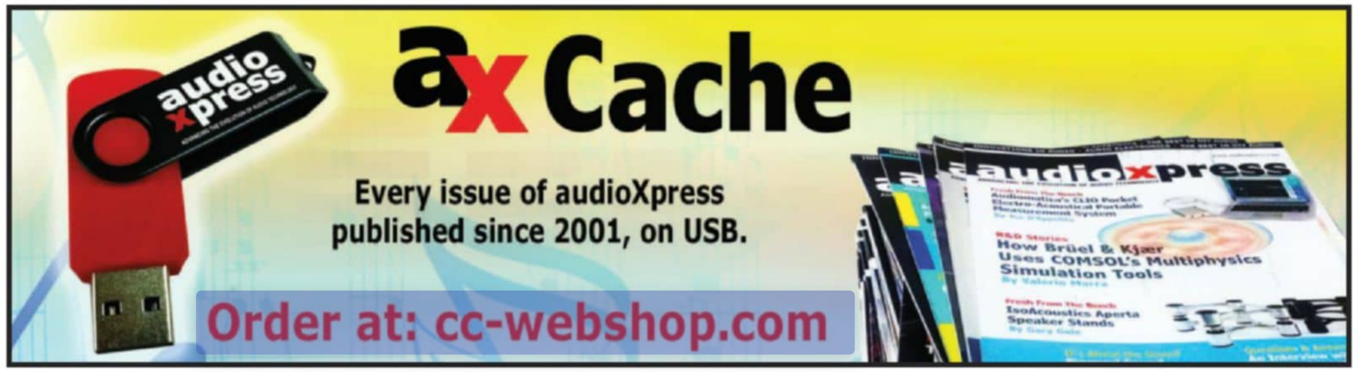

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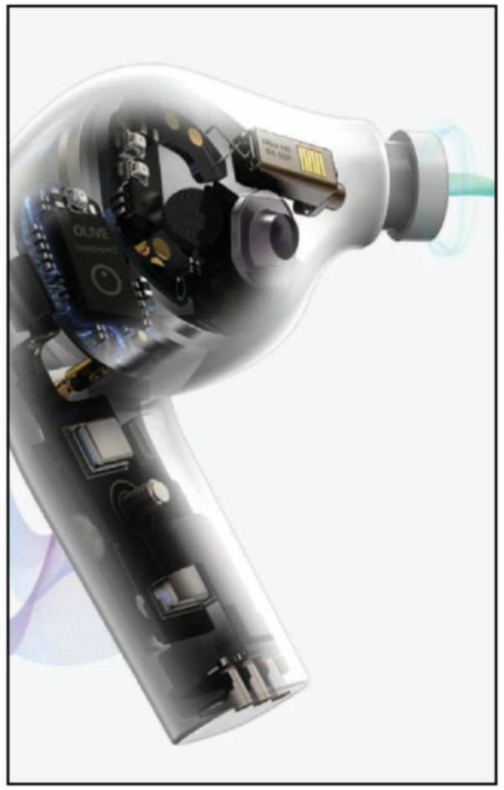
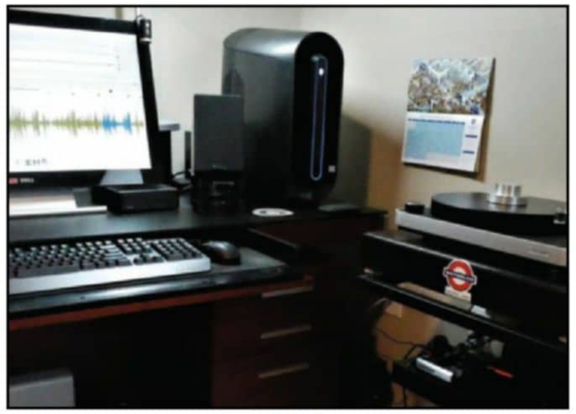
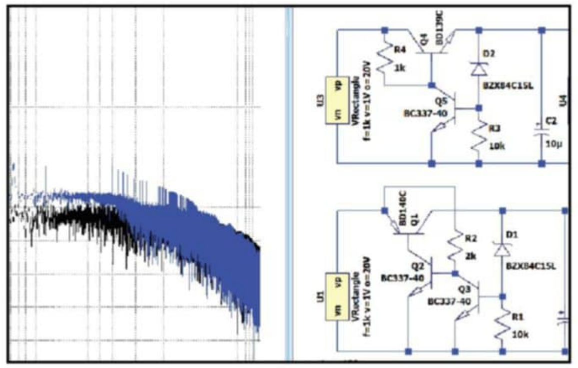

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Constantly Evolving Technology

By
Mike Klasco and J. Martins

There's a lot happening in speaker technology. Drivers and transducers are constantly evolving, and innovations are happening not only in design, construction, and materials but also in the engineering approach. In this report we will focus once again on innovations in speaker technologies.

Loudspeaker designers can look forward to a couple of exciting materials that have been in stealth mode but are about to step into the spotlight. Iron Nitride is an example of a new permanent magnet material with a formulation that breaks the compromise between ferrite and neodymium, and for manufacturers in particular, represents an option that gives neodymium a run for its money. Next up is Hiperco 50, a soft magnetic alloy that is a higher performance alternative to low carbon steel for lighter, more compact, and more powerful magnetic structures.

And we can add less traditional approaches to improve specifications by adding metamaterials, the latest option to increase the virtual back volume or more effectively absorb sound energy.

Let's Start with the Magnetics

Neodymium Iron Boron (NdFeB, or neo) is a hot topic, but for the wrong reasons. The news is not about innovation but rather adaptation to a "new normal" for the price of neodymium (neo for short). Significant cost inflation is expected to continue as demand from electric vehicles, wind turbines, and all other magnetic applications compete for limited supply. The usual supply chain bottlenecks aren't going away anytime soon and geopolitical tension has contributed further to heighten prices and supply risks. Downgrading to ferrite comes with compromises to size, weight, and performance—often a non-option.

Thankfully, just on the horizon is the first new permanent magnet material for speakers in decades. Niron Magnetics has developed technology that enables permanent magnets to be fabricated from Iron Nitride. The result is a powerful permanent magnet that does not utilize any rare earth elements. Iron Nitride promises several major advantages over traditional rare earth magnets, including enhanced temperature stability, price stability, and supply assurance.

Sustainability is another part of the story, with ~75% less CO_2 emissions over the life cycle vs. neodymium, providing a path for organizations to hit emission reduction targets, and presenting an opportunity for product differentiation with the growing segment of eco-conscious consumers.

Unlike neodymium or cobalt, Iron Nitride is composed of commodity ingredients, readily available around the world. Insulated from geopolitical factors and globally manufacturable, Iron Nitride stands to minimize the impact of tariffs and logistics on speaker manufacturers.

Niron Magnetics was born out of the labs of the University of Minnesota, with initial funding from a Department of Energy grant and later rounds of funding from Volvo and other big companies—about $34 million in financing behind the company to date. Today, its work on Iron Nitride is led by the former head of Magnetic Materials R&D at General Electric. Using a patented, scalable process, Niron Magnetics is developing the first mass-production process for Iron Nitride permanent magnets. While the ultimate goal is to produce magnets with the performance characteristics required for electric vehicles (EVs), earlier in its product pipeline are Iron Nitride magnets for the speaker industry. The road map projects that samples will be available for Niron's audio design partners late in 2022.

Niron's first generation of magnets is intended to break the compromise faced by speaker manufacturers today, representing a significant step up from ferrite's performance at a lower and more stable cost than neodymium. The initial energy product of Iron Nitride magnets will be between that of ferrite magnets (which are below 5 MGOe) and sintered NdFeB magnets (ranging from about 30 MGOe to 50 MGOe). This could be the solution for speaker designers and manufacturers who have reluctantly downgraded from neodymium designs to ferrite designs. Cost and weight/size sensitive applications stand to benefit significantly as well as markets that have long-term contracts, where volatile input material costs pose a serious business risk. The other potential winners are designers for applications where temperatures are higher (either environmental

Niron Magnetics is the company developing the world's first high-performance Iron Nitride permanent magnets, free of rare earths. Niron's technology delivers magnets that are less expensive, more sustainable, globally available, and made from abundant input materials not subject to supply constraints or price instability.

temperatures or temperatures reached by hard-working speakers from concert sound to active noise-cancellation speakers in cars) that will benefit from Iron Nitride's intrinsically stable performance at high temperatures.

In 2024, Niron Magnetics plans to introduce Iron Nitride magnets that directly compete with sintered NdFeB magnets. Similar to neo displacing AlNiCo in the 1980s, integrating new material into speaker designs favors companies committed to moving quickly on innovation. For the companies that are willing, Iron Nitride magnets promise a competitive edge—an answer to many of the intractable problems faced by speaker manufacturers today, with new benefits to boot.

Soft Iron Alloy

With high-energy product magnets (e.g., neodymium and Iron

Born from the original Philips Speaker systems, with R&D facilities in Belgium, US, China, and Malaysia and manufacturing facilities in Belgium, Hungary, China, Malaysia, and Mexico, Premium Sound Solutions has become one of the world's leading companies in automotive and consumer sound products all around the globe. The image shows a patented Coil Disc Drive (CDD) high-power woofer technology using an inverted magnet system to reduce the overall speaker height.

Nitride), the limitations of the magnetic return circuit begin to get in the way. Soft iron alloy's superior material for magnetic return structures has been around for decades but not on most of the audio industry's radar. Today if you tear down a balanced armature earphone driver or Apple's headphones and earphones, you won't find a low carbon steel magnetic structure but rather a soft iron alloy that is thinner, lighter, and has higher magnetic flux capacity.

Iron has been used for more than a millennium for tools (and weapons) and by adding a bit of carbon, a steel alloy was created allowing a disproportionate amount of strength to be gained. But for carrying magnetic flux, only enough carbon is needed to enable practical forming techniques. It is low carbon (soft) steel that is the default loudspeaker magnetic structure backplate and top plate material, typically cold-forged to shape. Steel 1010 alloy has 10% carbon, most cold-forged steel backplates are 1008 (8%). While top plates can be 1004, they are a bit soft to work with and are expensive and hard to obtain in the US—but readily available in Asia without a price penalty.

For almost a century there have been high-performance iron alloys used for mission-critical applications—and now with potent magnets (e.g., Iron Nitride and neodymium), these iron alloys offer real performance benefits for magnetic structures from compression drivers to headphones and balanced armature earphones and hearing aid transducers. Enabling thinner microspeakers for smartphones, lighter and shallower auto sound speakers, lighter and smaller earphone and headphone drivers, compression drivers with higher Tesla voice coil gaps, and more. While not always cost-effective, most electro-dynamic transducers would benefit from low magnetic flux resistance soft metals.

It all started with Permendur, invented in 1929 at Bell Telephone Labs, a Cobalt Iron soft ferromagnetic alloy with equal parts of cobalt and iron, notable for its high magnetic saturation (between 2.3T and 2.4T). Permendur presents a high permeability and low core losses and a low coercivity. Utilizing Pemendur in the pole pieces of speakers it's possible to reach beyond 2T (or 20Kg) in the voice coil gaps. The advantage of high saturation in a magnetic core is that it can function at higher magnetic field strengths, so it is more compact and lighter for a given magnetic flux and power level.

Supermendur, which some of us know as Carpenter's "Hiperco 50," was invented in 1957 and has a similar composition to

Carpenter's "Hiperco 50" was invented in 1957 and can be found today in multiple audio applications, including balanced armature earphones and hearing aid drivers.

Permendur 49 but is grain-oriented. Hiperco Alloy 50 can be found today in balanced armature earphones and hearing aid drivers.

An interesting article by Aminul Md Mehedi (Carpenter Technology) about the advantages of Hiperco alloys in microspeakers can be found online on the *audioXpress* website (see Resources). As mentioned, Hiperco soft magnetic alloys enable dynamic driver units optimized for increased performance, with electromagnet size reductions and decreased weight.

Acoustic Metamaterials

Acoustic metamaterials are now increasingly an additional tool in the designer's toolbox, even though they have been behind the scenes—relegated and buried in the speaker sub-enclosures of smartphones, and more recently finding its way behind compression driver and tweeter domes.

In a recent article in *Acoustics Today* (a publication of the Acoustical Society of America), Michael Haberman of the Applied Research Labs of University of Texas stated, "The defining property of a metamaterial is that it achieves effects not found in nature as a means to address long-standing engineering challenges in acoustics." And he adds, "The common theme of the many Acoustic metamaterials devices is an apparent defiance of the intuitive laws of physics, which often require strange concepts such as negative density and negative compressibility."

Acoustic metamaterials are the catalyst to dissolve Hofmann's Iron Laws, stating that a designer who wants good, deep low-frequency sound and high sensitivity can obtain these goals, but they will have to use a large speaker enclosure. Similarly, if a designer is forced by space constraints to use a very small cabinet, and they aim to get good, deep low-frequency sound, the sensitivity will be compromised.

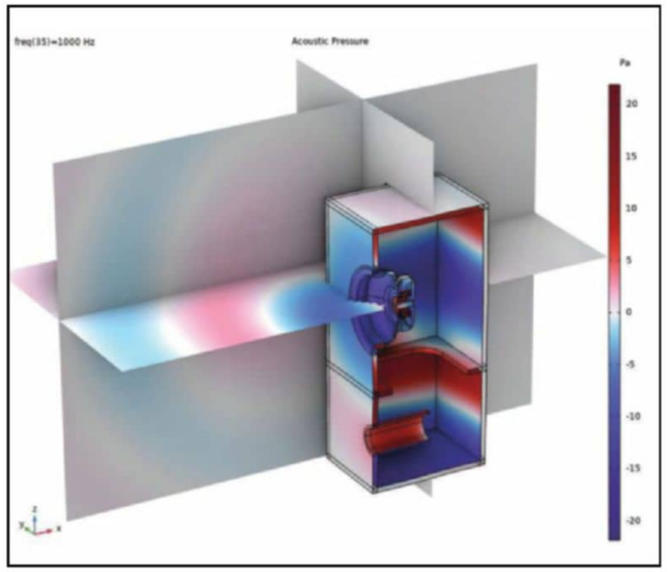
It was 10 years ago that Knowles introduced N'Bass (for "enhanced bass") Virtual Back Volume technology, a porous powder metamaterial filler for the rear cavity of its microspeakers (supplied to Apple). This super compliant/super porous box stuffing is more compliant than the air it displaces and increases the back volume seen by the speaker by up to 100%. It provides a combination of better acoustic performance—specifically more bass—or smaller

Metamaterial Absorption Technology (MAT) is a joint development initiative between KEF and Acoustic Metamaterials Group (Hong Kong) that resulted in a structure that has proven to eliminate 99% of all the unwanted sound that radiates from the rear of a driver. Currently implemented in the latest generation LS50 Meta loudspeakers.

Knowles' original N'Bass Virtual Back Volume Technology, a porous powder metamaterial filler, allowed "enhanced bass" in miniaturized speaker systems. This is a special material that increases the back volume seen by the speaker by up to 100% thus enabling either a better acoustic performance—specifically more bass—or smaller speaker box designs. It also enables the usage of bigger speakers with the same gross application volume.

loudspeaker box designs while keeping the same performance.

N'Bass also facilitates the usage of bigger loudspeakers in what would be impractically small enclosures. Similar to a sponge, 2 grams of the N'Bass material have a surface area of about a football field. The small back volumes used in mobile devices (due to space constraints) increase the air spring stiffness and this back pressure, raising the corner frequency of the turnover bass roll-off in the bass. The material desorbs the air molecules when the membrane vibrates out of the speaker box. This virtually increases the available back volume for the loudspeaker and since Knowles is the only supplier that offers this specific technology for miniaturized speaker systems,



The use of computer modeling and simulation software, such as COMSOL, is enabling a better understanding of advanced materials and even allowing the engineering of metamaterials optimized for speaker design.

it has been great for Apple. But that left out not just designers from other brands, but also applications (e.g., smart speakers, soundbars, and subwoofers). Then in 2016 Knowles sold its microspeaker group along with the N'Bass technology to Sound Solutions International in China. What is now making N'Bass relevant is this captive in-house technology has been spun off as SSI New Material, specializing in nano metamaterials and happy to hear from new customers and applications. And since N'Bass technology has been continuously improved, it has been successfully applied to other product categories, making them thinner and lighter without compromise of acoustic performance.

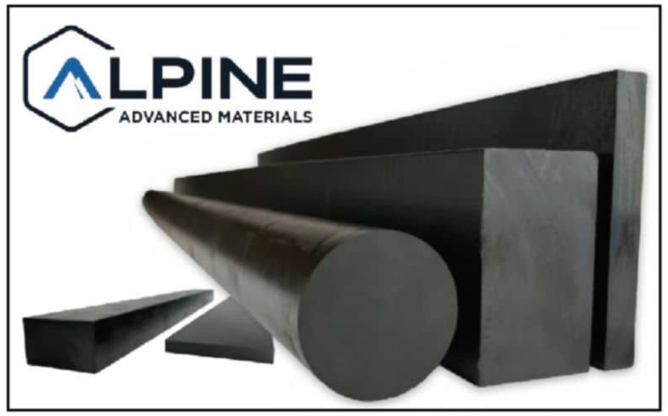
But Wait, There Is More

Another challenge that acoustic metamaterials have conquered is to create a metamaterial acoustic absorber anechoic back chamber in a small space that defines conventional physics. Acoustic metamaterial absorbers can realize previously unattainable absorption spectra with sub-wavelength dimensions approaching the theoretical minimum.

KEF recently collaborated on a back chamber enclosure for a tweeter with Acoustic Metamaterials Group (AMG), with AMG providing the design and geometry of the metamaterial structures. Check out the Audio Engineering Society (AES) preprint by Seabstien Degraeve and Jack Oclee-Brown "Metamaterial Absorber for Loudspeaker Enclosures," AES 148th Convention (www.aes.org/e-lib/browse.cfm?elib=20758)

Behind the tweeter dome of KEF's LS50 Meta coaxial monitor speakers reside KEF's Metamaterial Absorption Technology (MAT). A highly complex maze-like structure that eliminates reflected energy from the rear radiation of the diaphragm, MAT's tuned channels absorb 99% of the unwanted sound from the rear of driver, compared to around 60% absorption from loudspeakers using different approaches. In the AES paper, they explain that at higher frequencies acoustical cavity resonances occur and these may lead to frequency response aberrations as a result of irregularities in the loading impedance experienced by the driver. Acoustical wadding (usually a felted formed dome directly behind the dome) or midrange/tweeters into the back chamber help to reduce the amplitude of the cavity resonances. The wadding quantity must be sufficient to prevent any aberrations in the frequency response, but an excess is not desirable either as the low-frequency bandwidth is reduced.

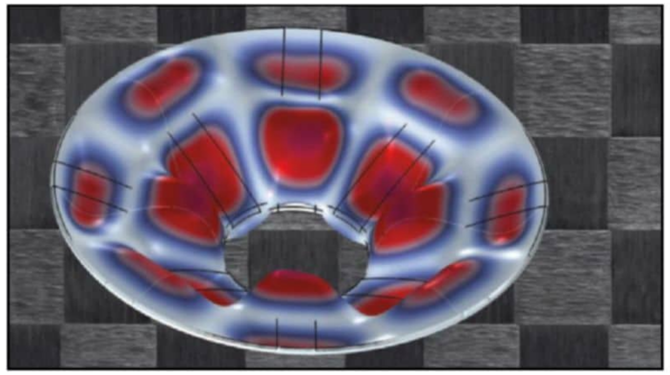
To shrink the size of the rear absorber, one possible approach is to use resonant acoustical absorbers. such as Helmholtz oscillators or quarter-wavelength ducts and lossy transmission lines (think Bowers & Wilkins' Nautilus). Acoustical resonators are extremely efficient subwavelength absorbers, but they have the disadvantage of being narrowband in character. Adding damping increases the bandwidth but limits the absorption. An ideal solution would be to arrange a structure containing many high-Q resonators optimized to provide a wide bandwidth of overall absorption. In this context, acoustic metamaterials can deliver unconventional effective properties without the constraints normally imposed by nature.



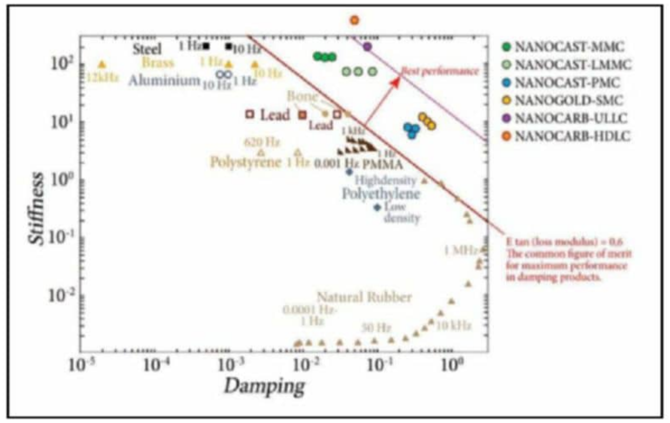
HX5, a lightweight military aviation-grade nanocomposite from Alpine Advanced Materials presents an alternative to beryllium or high-performance aluminum alloys for compression driver and tweeter domes.

With HX5, the injection-molded prototypes can be affordably extruded or machined, and tested to anticipate the performance of the finished product.

This studio monitor from Brooklyn, NY-based Ex Machina Soundworks features custom SEAS drivers with TeXtreme midrange cones in sealed Valchromat cabinets. Commenting on the development, Dev Avidon, founder and CEO of Ex Machina Soundworks says that the requirements for this coaxial drive would never be achievable without the tight control of break-ups made possible by Composite Sound's approach of engineering the thin-ply carbon diaphragms



Inspired by the metamaterials approach, Composite Sound developed its new Composite Sound Metamodal technology, which controls and manipulates resonances in a speaker diaphragm by optimizing the thickness and stiffness tri-dimensionally, achieving tailored engineering of modal behavior.



Netherlands-based material company Diluvite developed a series of super stiff and high damping Nanocast composite laminates focused on vibrational control with multiple applications on speaker design. Dilutive developed a specially structured crystalloid metal matrix that the company claims to provide better damping loss capacity and than any commercially available alloy, with excellent thermal conductivity, comparable to aluminum alloys, which means it can be used as a heat dissipation material (heatsinks) in advanced loudspeaker cabinets and chassis.

With a metamaterial, the required macro-properties are synthesized by creating a highly customized miniaturized structure. Typically, but not exclusively, a small unit-cell structure is repeated to create a larger region of material having highly tuned properties. The absorber contains a continuum of acoustical resonators, tuned to have a constant number of resonances per octave above a cut-off frequency, showing that the metamaterial absorber behaves like a compliance. Below and as expected, the metamaterial is just a pure compliance related to the physical volume of the system. Through customization, we can achieve broadband low-frequency noise absorption better than any material found in nature.

Advanced Materials and Processes

Speaker engineers always have an appetite for new cone and diaphragm materials. Thermoplastic composites have great potential, but most of these have been in the form of sheets for compression molding using a positive and negative die. The challenge has been getting a deep enough draw for forming shaped parts, and the even greater challenge is the audio industry is very much invested in injection molding, not compression molding.

The latest entry in thermoplastics is HX5, a lightweight military aviation-grade nanocomposite from Alpine Advanced Materials (Dallas, TX). Developed by Lockheed Martin Skunk Works (Palmdale, CA), and with more than a decade of testing and validation with an R&D investment in excess of $50 million, the new material presents an alternative to beryllium or high-performance aluminum alloys for compression driver and tweeter domes and has high-temperature tolerance. Compared to aluminum, domes can be made at half the weight.

While Alpine's HX5 does require changing the manufacturing process, it enables designers to convert from metal to a material that allows further gains in production and improve the quality of finished products. This combination allows manufacturers to achieve not just more affordable designs but superior prototypes, even when injection molding is involved. While cost-effective for production, injection molds aren't normally efficient for prototyping and are only viable after a design is finalized. With HX5, the injection-molded prototypes can be affordably extruded or machined, and tested to anticipate the performance of the finished product.

Alpine is also available to help on the conversion process, with the advantage that the use of finite element analysis (FEA) and simulation will be more directly reflected in the prototype stage by directly controlling the design topology optimization even at the detail of fiber orientation. Something that is proving increasingly effective in loudspeaker design.

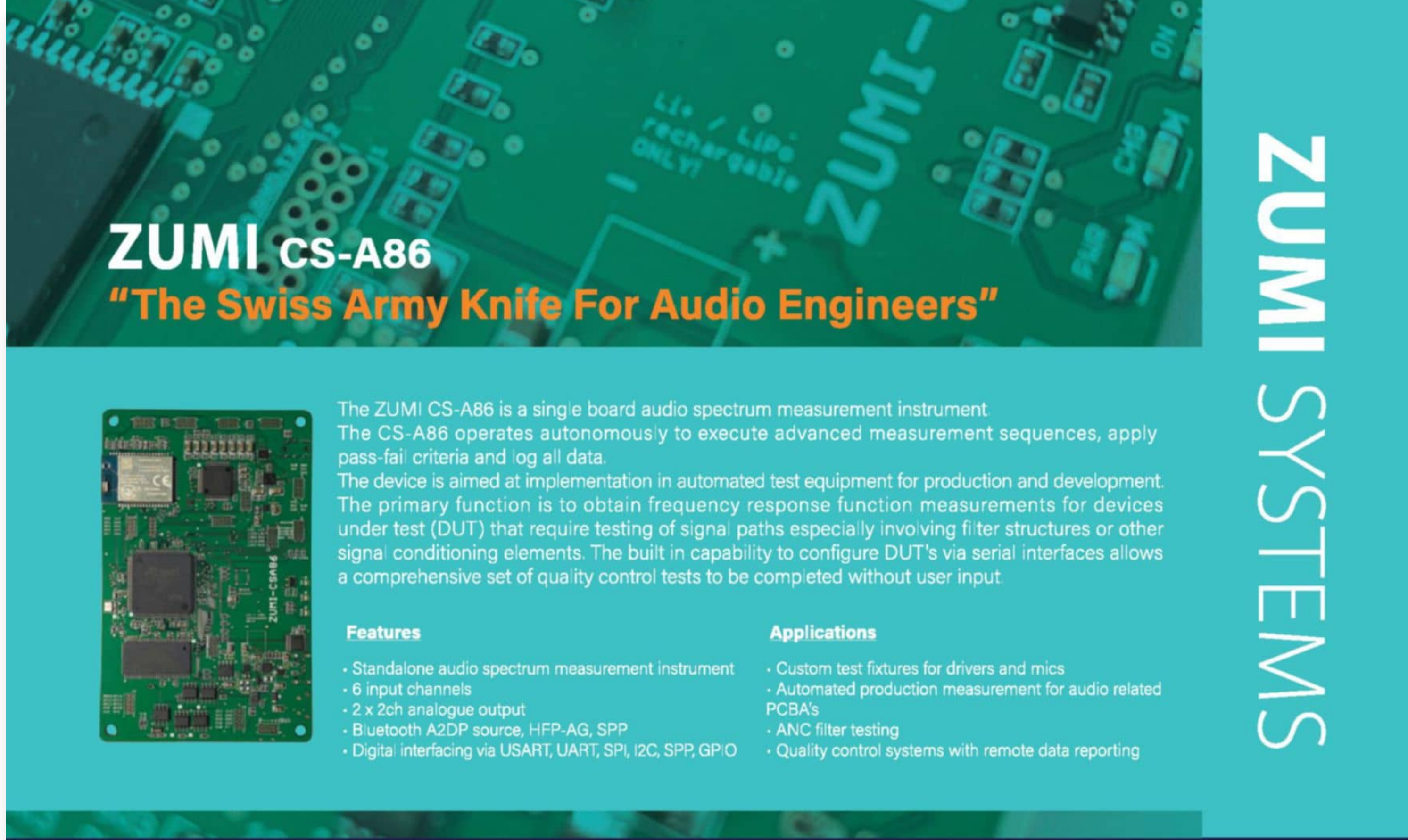
Independently on the material choice and how the cones, domes, or membranes are formed, this is an approach that is proving effective to generate completely new levels of speaker optimizations.

Tectonic is an example of a company using this approach extensively to optimize the composite honeycomb sandwich material used in its Balanced Mode Radiator (BMR) drivers. The BMR topology provides a full-range drive response that has a similar acoustic behavior to a point source drive unit where the majority of the audio band is produced from a single membrane. A BMR is a modal loudspeaker that uses a flat membrane, where a limited number of bending resonances are harnessed to produce midrange and high frequencies. The ability to fully model and simulate the behavior of this membrane, which is subsequently formed to the same exact specifications that were generated by the FEA software tool by using new materials and formulations, opens new possibilities that Tectonic is fully exploring in its latest generation of BMR products (see text on p. 19).

The result is a cone diaphragm with very tightly controlled break-up, with no peaks and dips in a wide frequency response.

Metamodal Engineering

In previous speaker articles published here in *audioXpress* and its sister magazine *Voice Coil*, we have discussed a new category of Thin-Ply Carbon Diaphragms (TPCD, made from TeXtreme carbon),



Pictured are the high-end Aequos Audio Adamantis speakers, which use Diluvite's Nanocast materials in its cabinets.

introduced by Oxeon in Sweden, an advanced composites fabrication specialist that holds patents both for its thin-ply and spread tow technologies and unique materials.

Originally developed for aerospace, industrial, and sports applications, Oxeon perfected TeXtreme, quickly turning it into a sought-after solution for the most demanding applications that require ultra-light weight, high stiffness, and long endurance. TeXtreme technology is now being used by Formula 1 teams and by NASA for its Mars drone, as just two outstanding examples.

A thin-ply carbon material is created by arranging layers of carbon fiber in an optimized manner. The fibers in these layers can be oriented in different directions, creating a complex 3D weaving structure as required by the application. By optimizing the way the layers are organized, the mechanical performance can be maximized, the weight minimized, and the part can be engineered to behave in certain desired ways, in different areas and different directions.

The use of TeXtreme was an immediate success in the loudspeaker industry, with Eminence and SB Acoustics launching tweeters, compression drivers, midranges, and bass/midwoofers

Resources

Acoustic Metamaterials Group | https://acousticmetamaterials.org

Alpine Advanced Materials | https://alpineadvancedmaterials.com

Composite Sound | https://composite-sound.com

Dutch & Dutch | www.dutchdutch.com

Hiperco Alloy 50 | www.carpentertechnology.com

A. M. Mehedi, "Better Microspeakers Utilizing Hiperco Alloys," *Voice Coil*, November 2020, https://audioxpress.com/article/better-microspeakers-utilizing-hiperco-alloys

Niron Magnetics | https://nironmagnetics.com

Output, Inc. | www.output.com

Pantheone Audio | www.pantheoneaudio.com

Sound Solutions International | www.sound-solutions.com

Syng Space | www.syngspace.com

Groundbreaking, beautiful, and inspiring are just a few ways to describe the first design from Pantheone Audio, a winner of the prestigious Red Dot Award for design and an Archiproducts Design Award. Created upon the three pillars of art, form, and sound, the Pantheone I state-of-the-art sound system is made from handmade high-density resin, and offers not just a desirable sculptural piece to have within the home, but uncompromised sound and technology capabilities.

based on this technology, while multiple brands are already exploring its performance and advantages very enthusiastically (Perlisten Audio and Ex Machina Soundworks are two notable examples, and more are on the way).

The adoption of TeXtreme and thin-ply carbon diaphragms motivated the creation of a dedicated company to lead and expand these efforts, which led to the recent founding of Composite Sound, which is a spin-off from Oxeon.

Composite Sound engineers TPCD to prevent undesired resonances from ever occurring. Where traditional speaker diaphragms try to limit the negative effects of undesired resonances, the company's process optimizes the thickness, stiffness, and damping in different parts of the diaphragm, controlling modal behavior. For now, Composite Sound main focus is on the development, process, and fabrication of cones, domes and diaphragms for loudspeakers, but the same technology can be beneficial in other applications in acoustics and other parts for audio products.

Composite Sound uses this material science combined with advanced simulation tools and a unique engineering approach to create its speaker products. By expanding the approach to fabrication, they are able to maximize stiffness and minimize weight without compromising the controlled behavior.

And Composite Sound is now announcing a more advanced, second-generation solution with its Metamodal technology. It builds on TPCDs with distributed break-up modes and takes it to a whole new level by enabling cones and diaphragms with engineered modal behavior.

The term "Meta" (as a prefix for beyond/transcending) describes something that goes beyond the naturally occurring. Composite Sound Metamodal cones and diaphragms are engineered for a performance level and specific behavior, which is beyond the naturally occurring material properties that bounds the performance of traditional speaker diaphragms.

In the same way that metamaterials can be artificially structured to control light, and metasurfaces are able to manipulate soundwaves, Composite Sound Metamodal technology controls and manipulates resonances by optimizing the thickness and stiffness in all areas of the diaphragm. The Composite Sound Metamodal diaphragm can be engineered for a tailored modal behavior—frequency response linearity, improved pistonic motion, and moving mass optimization—as needed.

Traditionally, speaker cones, domes, and diaphragms offer either increased stiffness to push the break-up frequency higher, or increased damping to reduce peak amplitude. The Composite Sound Metamodal approach engineers the modes so that there are no severe peaks to move or reduce. Stiffness can be focused to the right places for increased performance, while thickness can be reduced where it is needed less, thereby reducing mass. Compared to the first-generation TPCDs, Composite Sound Metamodal diaphragms show increased break-up frequency, reduced weight, and even more controlled break-up behavior.

For developers working on new designs, this tailored metamodal approach also means the ability to fully optimize a driver to an industrial design or meet challenging requirements in terms of weight, size, and efficiency for a higher performance product.

The possibilities of engineering a cone to an exact intended behavior also paves the way for manufacturers to upgrade existing designs with a new diaphragm that directly replaces an existing part with the same exact intended "sound signature," without the nonlinearities that limited its original performance. It also allows for better acoustic performance—specifically more bass—or smaller speaker box designs. It enables the usage of bigger speakers with the same gross application volume.

BDNC

Flat Glass-Composite Speaker Drivers

When addressing speaker materials and innovations in drivers, it's important to mention BDNC (Holding) Ltd., which offers its highly advanced flat glass-composite diaphragm woofer. This new innovative family of drivers uses ultra-thin tempered glass as the diaphragm, combined in a low-profile woofer able to produce extended bass response. Available in a large 584mm×92mm×25mm size, plus 80mm×20mm×7.2mm and 160mm×40mm×17mm models, the company also offers headphone drivers using the glass-composite technology.

BDNC derives from Beijing Design and Creation Electronics Co., Ltd., founded in Beijing as an algorithm provider for semiconductor companies. Following a management buy-out (technology and assets) by the company founder in 2007, BDNC successfully raised the capital needed to pursue its separate research activities and was finally registered in Taiwan. There, the company established technical facilities to expand its business from digital signal processing (DSP) and mixed-aignal research, and also into electro-acoustics, product development, production, and commercialization.

In 2016, BDNC introduced the results of an extended R&D effort focused on using ultra-thin tempered glass, enabling the company to successfully introduce a new generation of flat thin speakers able to generate extended bass response.

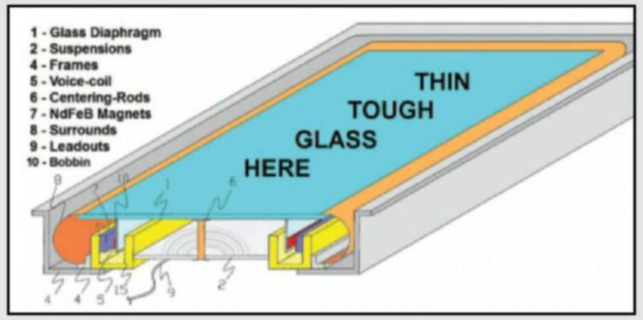
The use of thin glass (amorphous SiO_2) as the sound radiating diaphragms for these drivers' designs improves the performance of traditional designs. The company's glass diaphragm is currently available with a thickness from 0.8mm down to 0.03mm, allowing it to be used as a flat or (traditional) cone diaphragm, in standalone or composite form.

Proven advantages include improved speed of sound propagation in shear mode, density (impacting directly speaker efficiency), mechanical strength (stiffness, Young's modulus, shear modulus, elongation before break, degree of bending before break, yield strength, creeping properties, fatigue properties, etc.), and internal loss factor (natural sounding).

The material also shows strong advantages in terms of minimum production and working thickness, facilitating mass-production, and adhesion properties, reducing the production cost and yield, while also showing improved thermal properties characteristics.

www.newbdnc.com

The BDNC Flat Glass Speaker uses Schott AS87 Glass in a composite with a viscoelastomic damping layer and aluminum alloy foil to create a flat diaphragm just 0.3mm thick, and achieves an impressive 8mm excursion (Xmax peak-to-peak).



Actual implementation of the glass-composite-diaphragm woofer with details of the patented design.

Sound Solutions International

Addressing the Unique Challenges of Micro-Acoustics

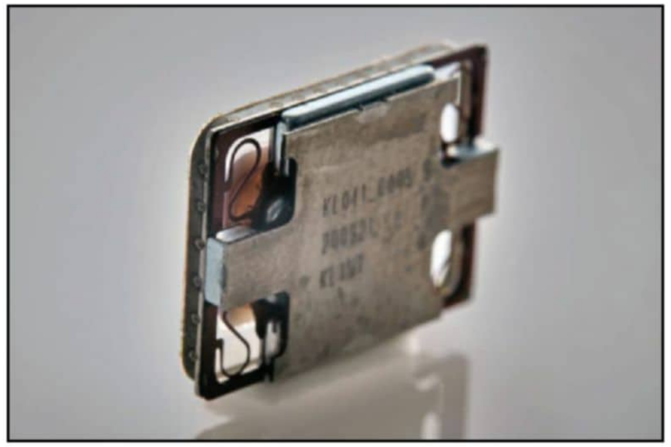
Sound Solutions International (SSI) was founded in 2016 but has a long heritage of more than 90 years in developing and manufacturing acoustic components and solutions. SSI is the successor organization of the former Philips Speaker factory in Vienna, Austria, where it operates as a global competence center for mobile acoustic solutions and the company's development and innovation center. For three decades, the company has led the acoustic component market for mobile devices under different ownerships from Philips, NXP Semiconductors, and Knowles Electronics. Since October 2021, Sound Solutions, headquartered in Zhenjiang, China, is part of the global Foxconn Group FIT (Foxconn Interconnect Technologies).

The audio quality of smartphones, even when the speakers are low distortion, is still marginal due to the limited dynamic range and weak low-end (700Hz at best). Equalization won't help inadequate excursion, and even when the suspension and mechanical clearance is provided, the peripheral one-point suspension is not stable enough to suppress rocking. Now SSI has introduced its latest microspeakers with a unique and patent-pending coil anchor system (CAS) to reduce rocking on the short and long axis and to gain additional sound pressure level.

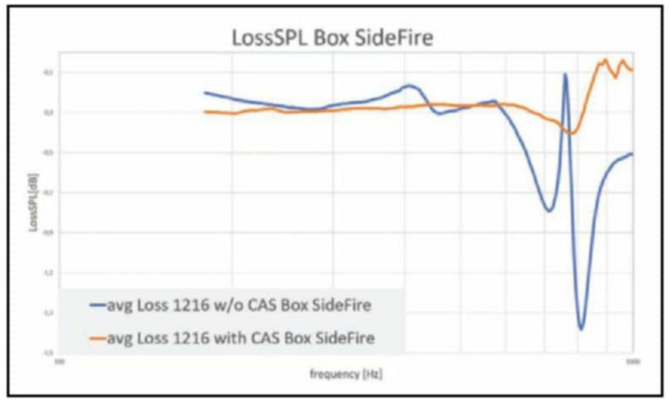
Han-Peter Mutzel of SSI explains that the ideal mode for electro-dynamic speakers is the piston movement of the membrane, which means that maximum possible air shifting can be achieved. But the physical effects (e.g., Eigenmodes) of the moving system let the membrane "rock," reducing the shifted air, which results in lower sound pressure and increased distortion. SSI's solution is to shift the so-called rocking frequencies into an area where they do not harm the acoustic performance by adding a second suspension system (CAS) to "anchor" the coil.

Four spring elements in the corners increase the lateral stiffness on both the long and the short axis of the speaker. Besides the rocking suppression, the spring elements are the contact system for the coil itself. This means SSI has been able to remove the lead-out wires and gain extra space for more magnet material. This not only eliminates a quality risk of breaking wire loops but also enables a higher electrical efficiency of the speaker.

www.sound-solutions.com



The new generation of Sound Solutions CAS speakers is built on a highly scalable platform, allowing the company to react quickly to different requirements. It can be easily scaled on size, but also on performance with flexible multi-magnet systems.



To measure the impact of rocking modes on the performance, SSI monitors sound pressure levels (SPL) defined as "LossSPL." An ideal speaker's piston mode would show a 0dB straight line over the whole frequency range for that parameter. The measurement graph shows a significant SPL gain, achieved by the CAS design. The rocking behavior is heavily impacted by the application, especially by asymmetric air-load like side-firing sound ports. The graph shows that the SSI solution is robust against such effects, which enables leveraging the full excursion headroom even in highly restricted applications.

Sound Solutions International (SSI) introduced its latest family of high-performance microspeakers featuring the company's Coil-Anchor System (CAS) rocking suppression technology. The patent-pending high-excursion Manticore speaker family already includes three models for smartphones and laptop/tablet applications.

Tectonic

New High Aspect Ratio Panel BMR Speakers

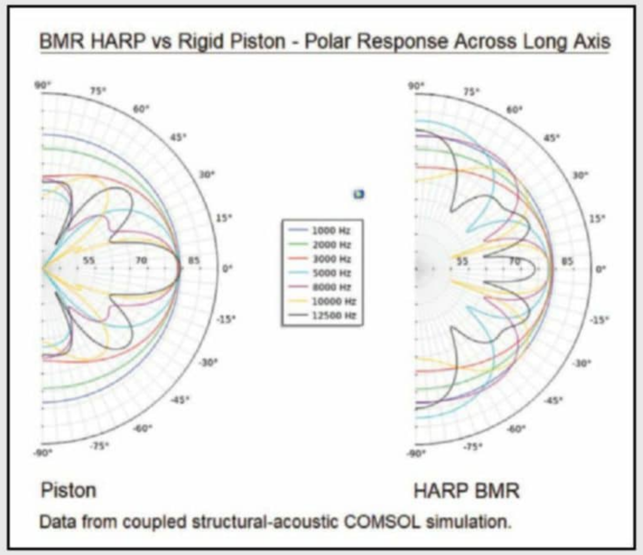
Tectonic has been around for a while, but the focus for the company has more recently been on continuing to expand its development of Balanced Mode Radiators (BMR), the next generation of their ancestor's NXT flat panel drivers. In recent years, the company has made significant inroads into large volume products that require the technical advantages of BMRs, particularly very wide directivity.

High-quality audio and innovative industrial design seldom coexist in perfect harmony, especially with today's consumer products becoming ever smaller, ever thinner, and packed to the gills with electronics. Tectonic plans to announce a family of High Aspect Ratio Panel (HARP) audio transducers at CES2022, for shipment mid-year 2022. The HARP loudspeaker is a slim rectangular BMR that achieves both extended frequency response and wider directivity along its length (long axis) and width (short axis).

The first member of the HARP family will be a narrow profile compact mid/high-range speaker. Especially handy for applications with constrained mechanical dimensions along with a unique replacement for conventional high-frequency (tweeter) components in two-way systems allowing for a lower (<1kHz) crossover point. Improving upon previous designs, including a highly compact form-factor and new suspension approach, the new Tectonic HARP measures only 100mm (L) × 16mm (W) × 20mm (H) making it highly attractive for TVs, monitors, A/B/C pillars in cars, compact soundbars, and more, where it could be easily crossed over to a compact bass drive unit. Tectonic is already working to launch a full-range variant, currently in development.

The intentional modal behavior of the BMR adds in superposition to the piston output, producing a substantially wider directivity above the frequency where the piston would be beaming. The BMR also provides a smooth sound power response ensuring high intelligibility and low coloration audio.

www.tectonicaudiolabs.com



Preliminary data shows a comparison of directivity over the long axis. Piston driver (left), HARP BMR (right)

Exploded view of the new Tectonic BMR High Aspect Ratio Panel (HARP) mid/high-range compact design, which provides a narrow profile.

Trulli Audio

The Power of Thin Drivers

Trulli Audio is a company that *audioXpress* has mentioned on previous occasions, recognized from its pioneering effort with ThinDriver designs, in development since 2008 under the Prescient brand. The company evolved to focus on solutions for efficient consumer designs with its TD38S 2″ thin speakers as a promising solution for portable applications, automotive near-field arrays, and personal sound zones, due to its combination of high thermal power handling while maintaining a small footprint and shallow depth.

The flat diaphragm topology, repositions and expands the voice coil to the juncture of the diaphragm periphery and the juncture of the surround. Trulli is also in the late stage of development of a new-generation 10″ woofer with a 7″ diameter voice coil with less than 3″ depth based on the same principles.

Recently, Trulli Audio launched a proof-of-concept product for its speaker drivers, the JAM5 High Fidelity Bluetooth Speaker, which retails for $199. Four years in the making, it was engineered to leverage the company's patented ThinDriver technology. Whether on its stand in the living room, or strapped to a golf cart, the JAM5 surprises for the powerful response and easy setup.

Since the JAM5 was unveiled by Trulli Audio and offered direct-to-consumer on the company's website, it received very positive reviews. The secret is in the ingenious Trulli TD38S 2Ω square driver with its high thermal power handling, while maintaining a small footprint and shallow depth. The 38mm square diaphragm confers over 20% more piston area than a round diaphragm and its configuration enables more spider corrugations with huge gains in excursion. In the JAM5, the two TD38S units are combined with two opposite passive radiators and a center tweeter. The unique dual-layered passive radiators are tuned to maximize low bass extension while ensuring sound quality remains punchy and accurate.

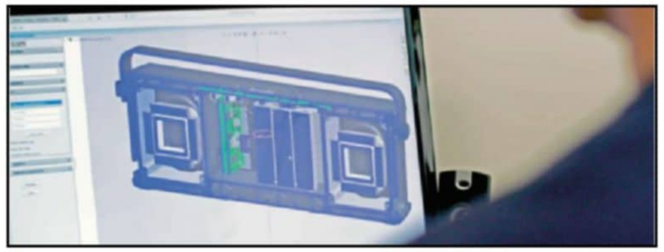
At the heart of the Trulli JAM5 is the ThinDriver, a multi-patented driver with an open ring of powerful n48 neodymium magnets around a large, perimeter driven voice coil, with a fraction of the weight and dimensions of traditional designs.

The Trulli Audio JAM5 portable speaker is rugged enough for the demands of outdoor life with a rating of IPX4, and is hand-assembled in the US.

While most speakers this size lack lower frequency response, with this driver design, the JAM5 delivers powerful and fast lows down to 60Hz. And another unique design aspect is the fact that the JAM5's housing features independent chambers for each speaker driver. Normally seen on higher-end tower speakers, this allows for more accurate tuning, stereo imaging, and frequency response. The combination of the center tweeter and ThinDriver perimeter speaker drivers deliver a broader spectrum of sound in a much smaller package than physics should allow.

Internal rechargeable battery not only allows up to 18 hours of listening (25% volume), but can also charge out to a phone or another 5V/1A USB-C equipped device. The speaker's volume goes all the way up to 91dB also thanks to its Class-D digital Smart Amp, which allows the JAM5 to be specified for a stereo peak output of 144W (72W per channel), with the response tune and optimized by in-line 48kHz digital signal processors. And since two JAM5s can be paired together for stereo listening, even outdoor listening is impressive.

www.trulliaudio.com



In the JAM5 design, the patented dual TD38 ThinDrivers, additional tweeter, and custom passive radiators work in concert with the enclosure itself to such an extent that the drivers are even adjusted to capitalize on the auditory effects that the all-metal grill has on sound output.

xMEMS

MEMS Microspeakers

Founded in January 2018, California-based xMEMS Labs is dedicated to designing advanced solutions and applications with MEMS technology for a range of consumer electronics devices. The company designed the world's first monolithic MEMS speaker, has already been granted 34 patents, and holds more than 100 patents pending for its technology.

Even though we will address headphone and earbud-related applications in our April edition, it is important to mention significant progress made on this front by xMEMS Labs, which confirmed that its first microspeaker—Montara—is now in production. Working closely with TSMC as its semiconductor foundry partner, the Montara design has passed all the required performance and reliability qualifications and has already been selected by manufacturers for future products, including upcoming true wireless earbuds designs, complemented with the company's companion Aptos Class-H piezo audio driver IC.

A monolithic MEMS speaker implements both actuation and diaphragm in silicon, resulting in an unmatched level of part-to-part consistency in frequency response and phase. Montara's fast mechanical response also results in the industry's lowest group delay and phase shift. For manufacturers, these characteristics reduce the need for calibration and speaker matching.

The 1mm profile, SMT-ready package and IP58 rating for dust/water simplify system design, integration, and assembly. Montara is qualified to both IEC, and the more stringent JEDEC standards for improved quality and reliability. This will allow consumers to benefit from waterproof earbud designs with improved audio fidelity.

New Smaller Models

Meanwhile, xMEMS has also introduced Cowell, its smallest model, targeting the replacement of balanced armature drivers in true wireless earbuds and hearing aid designs. At just 22mm³, 56 milligrams, and with a 3.2mm diameter side-firing package, Cowell delivers an impressive 110dB SPL at 1kHz, and is the first speaker to use xMEMS' second-generation M2 speaker cell architecture offering improved SPL/mm² enabling increased loudness in smaller form factors.

The new MEMS microspeaker from xMEMS is a much smaller and more efficient design, providing up to 15dB of gain above 1kHz for improved speech-in-noise performance and greater vocal and instrumental clarity versus electrodynamic and balanced armature microspeakers.

For true wireless stereo applications, Cowell can be implemented as a full-range driver in occluded earbud architectures or as a small, high-performance tweeter paired with an electrodynamic woofer driver in non-occluded or leaky two-way solutions. It is also a full-range driver that is 45% smaller than an equivalent balanced armature receiver, making receiver-in-canal applications a possibility. The side-firing package and 1mm-thin profile provides for placement flexibility in the earbud making space for larger batteries and additional sensor components. The SMT-ready package with an IP58 rating for dust/water simplifies system design, integration, and assembly.

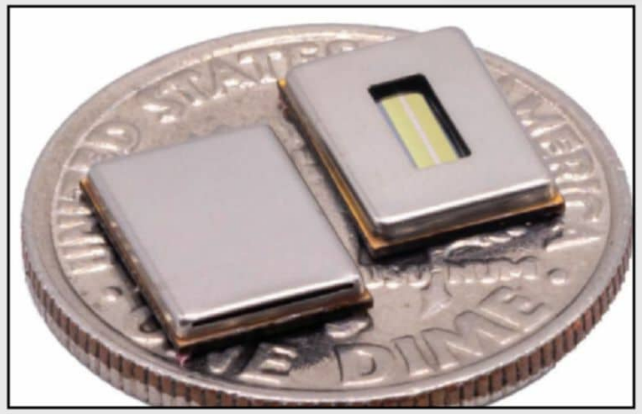
xMEMS Labs, a company founded just in January 2018, started mass production of its Montara monolithic MEMS µspeaker in 2021, partnering with TSMC, the world's largest semiconductor foundry. In the image, the Montara is shown with xMEMS latest and smallest driver for hearing aids and earbuds applications, the Cowell side-firing solution.

And just in time for this edition of *audioXpress*, xMEMS Labs unveiled the latest Tomales MEMS tweeter microspeaker with top and side-firing package options in a 1mm-thin profile that simplifies speaker placement and positioning to direct audio toward the ear in smart glasses and augmented/extended reality AR/XR headset applications. It has an IP58 rating for dust/water that simplifies system integration in these designs.

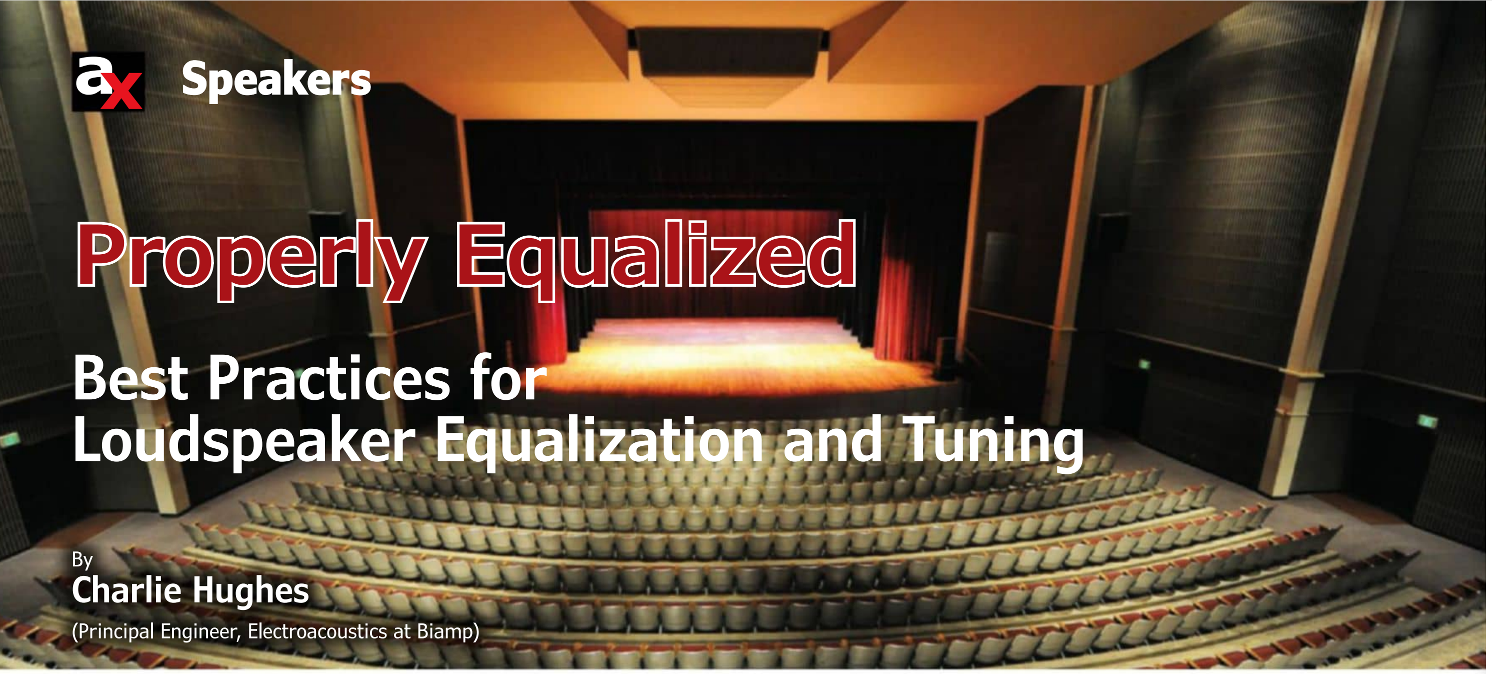
In free-air at 3cm, Tomales delivers 75dB SPL at 2kHz, 90dB at 4kHz and over 108dB at 10kHz. Tomales also implements xMEMS' second-generation M2 speaker cell architecture offering improved SPL/mm² enabling increased loudness in smaller form factors. In arrays, Tomales can work as a wideband tweeter. For CES 2022, xMEMS created an experimental application of a two-cell Tomales in a 16-chip array as a free-field tweeter in a two-way (woofer/tweeter) desktop speaker application.

Tomales samples and evaluation kits are available now for select customers, with mass production expected in Q2 2022. Tomales is available in both top-firing (6.05 x 8.4 x 1.15mm) and side-firing (6.05 x 1.15 x 8.4mm) LGA packages, and is paired with the xMEMS Aptos Class-H audio amplifier (1.92 x 1.92 x 0.6mm WLCSP).

www.xmems.com



The new Tomales MEMS tweeter from xMEMS Labs features a novel "bascule" design, allowing more opening, to gain additional displacement (higher SPL) in a smaller die size. These SPL levels also pave the way for interesting free-field applications in devices (e.g., laptops).



Properly Equalized

Best Practices for Loudspeaker Equalization and Tuning

By
Charlie Hughes
(Principal Engineer, Electroacoustics at Biamp)

A loudspeaker system should sound good and measure accordingly. This article offers some recommended guidelines that will not only help to develop good equalization and tuning practices, but also will go a long way in improving the overall sound quality, regardless of the application.

The tuning process begins with the loudspeaker. Before it even enters the space where it will eventually be installed, it is essential to measure the loudspeaker or loudspeaker system in a controlled environment. Ideally, this is somewhere outdoors where potentially problematic reflective surfaces won't be an issue. Understanding how the loudspeaker behaves naturally allows it to be preemptively tuned and optimized before it goes into a domestic listening room, home theater, or 50,000-seat stadium.

Often, these early, pre-installation measurements are pushed aside for the sake of time, but I firmly believe you don't have time not to do them. When skipping this all-important step, more than likely, more time will be spent on-site trying to get the system to sound right after it's installed. However, some problems might be difficult to diagnose, much less correct, due to the effects of the room and the boundaries impinging upon and becoming part of the measurement data. At a minimum, set up the loudspeaker in a parking lot, run some measurements, and preform some pre-optimization of the system. This should allow for more efficient and effective final tuning on-site.

Processing the Loudspeaker System

Once those reflection-free measurements have been captured at multiple locations, on-axis and off-axis, the next step is to apply the necessary processing (EQ, delay, gain adjustments, etc.) to the loudspeaker system. The goal is for it to sound good—regardless of what the measurements look like. If the measurements look good but the system sounds awful, something is wrong. It needs to sound good, even if the measurements might seem to indicate otherwise. It might be that something is amiss with the measurement setup. If the measurements don't look good, but the system sounds good, take time to explore why it's measuring that way.

After completing the pre-optimization and tuning to get the loudspeaker system to sound the way it should prior to loading it on the truck and getting it inside the building, most likely any issues encountered on-site during the installing and commissioning are not a problem with the loudspeaker. At this stage, poor results could be caused by several things: something changed in the rack (tuning settings, hardware, or wiring—each of which is easily traceable with electrical measurements); the acoustics of the venue; or boundary loading issues that need some further EQ adjustments in the affected frequency regions.

If everything was installed correctly (and it's been verified, don't assume) and the acoustics of the venue are in good shape, typically, boundary conditions might cause some issue in the low and/or midbass frequency regions. With the exception of some big stadiums, loudspeakers are usually mounted near boundaries in most installations. Mitigating boundary loading effects will require some corrective equalization in these frequency regions to account for the proximity of those boundaries.

Keep in mind that the only thing that can really be equalized is the direct sound from the loudspeaker, which needs to be as good and natural as possible. If the acoustical environment of the venue is such that it has a long reverberation and decay in the mid-bass region, say 200 Hz to 400 Hz, the sound might be muddy or tubby. There's going to be a desire to pull down those frequencies in that range. This is only a Band-Aid, but it may be the best option if the root cause of the problem (the acoustics) can't be addressed within the constraints of money and/or time. It certainly can be beneficial to keep things from sounding muddy, but there's a risk of pulling energy out of the source material in this frequency range. As a result, the audio isn't going to sound as natural as it could. This is the compromise to be made for not fixing the root cause of the problem. As shown in **Figure 1**, beamforming can be used to help to minimize the excitation of a venue with potentially problematic acoustics.

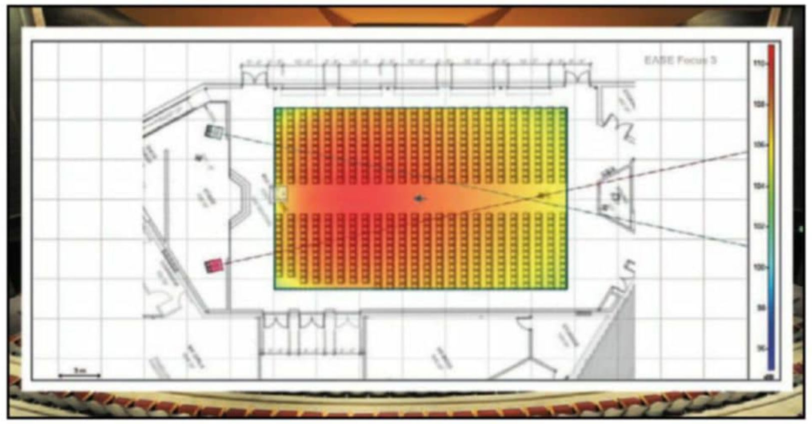


Figure 1: Beamforming provides the ability to combine multiple cabinets into seamless acoustic arrays with a myriad of potential coverage combinations. EASE software is a valuable design tool to evaluate and define coverage patterns and choose cabinet splay angles.

Multi-Purpose Venues

When working in a multi-purpose venue, there might be requirements for musical programming as well as speech. Sometimes the processing requirements might be a bit different for each of these if they are both to be optimized. Two different, parallel processing paths can be implemented to help with these types of applications. Generally, the music program should be able to use the tuning for the system. The speech signal path, however, might need additional processing to attenuate some of the

mid-bass and low frequencies and boost some of the midrange and vocal intelligibility frequencies, 1 kHz to 4 kHz. As an aside, some loudspeakers might not be as well behaved as others. A lack of sufficient directivity control in the 2 kHz to 4 kHz region can result in energy getting sprayed to undesirable locations. This is where well-behaved loudspeakers, such as the Community LVH-900 loudspeaker (**Figure 2** and **Figure 3**) from Biamp, can be incredibly beneficial.

Figure 2: This rendering shows Biamp's Community LVH-900 Beamforming Venue Horn, which is a large-format point-source loudspeaker that allows precise directivity control, individually or in arrays, to meet the sound requirements in any application.

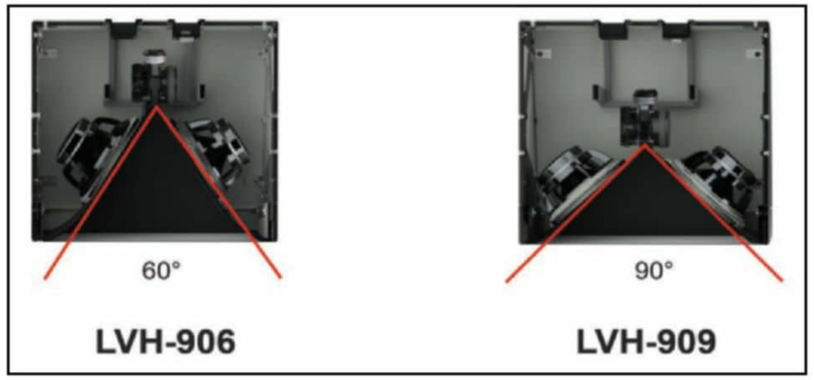


Figure 3: The Community LVH-900 offers two fixed horizontal coverages of 60° or 90°.

About the Author

Charlie Hughes has a Bachelor of Science degree in Physics from the Georgia Institute of Technology and has been involved in pro audio for more than 30 years. He is an active member of the Audio Engineering Society (AES), Consumer Technology Association (CTA), International Electrotechnical Commission (IEC), and Syn-Aud-Con as well as serving on AES, AVIXA, CTA, and IEC standards committees. In 2013, he received the CTA Technology Achievement Award for his standards work. He currently has seven patents issued relating to loudspeaker design, primarily dealing with directivity control.

Some venues might require front, under-balcony, or over-balcony fill loudspeakers. Time domain issues need to be properly addressed since it's not always frequency domain issues that can cause problems. The time alignment between the mains and fill loudspeakers should be such that, ideally, there is a seamless integration.

If someone is seated under a balcony, for example, their focus and attention shouldn't be drawn to a fill loudspeaker because of poor timing issues. Take advantage of the precedence effect, or the Haas effect. Slightly over delay those fill loudspeakers so that a listener doesn't focus on the fill loudspeakers, but more to the main or, even better, to the actual program on stage. In addition, if the fill loudspeaker is a full-range device, capable of reproducing low frequencies yet does not have good low-frequency directivity control, be very mindful of the rearward-radiated sound from that loudspeaker. Generally, some equalization is required to attenuate the low frequencies and mid-bass energy in the frequency region where the loudspeaker lacks sufficient directivity control. For someone seated closer to the stage than the fill loudspeaker, hearing low-frequency sound from behind them can be quite distracting.

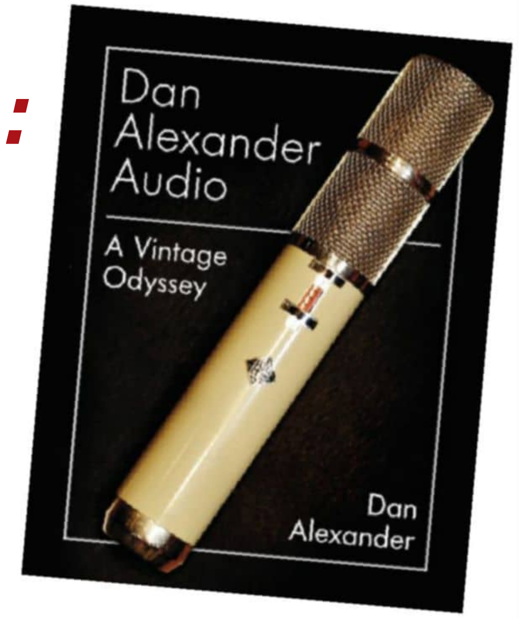
In Conclusion

Finally, listen—always listen. After tuning and alignments have all been completed (or so you think), walk around the venue. Listen to the mains from different locations. Listen in the overlap region between the mains and the fills. If it doesn't sound good, there's a problem and more work is needed. As mentioned earlier, measurements don't always tell the whole story. Instead of implicitly trusting metrics on the computer screen, it's essential to ensure the system actually sounds good throughout the venue.

How do you know when you've completed the tuning and commissioning process? Usually when the budget runs out. All jokes aside, it's a process that takes time to get just right—days, if not weeks, for large permanent installation applications. The loudspeaker system should sound good and measure accordingly. There shouldn't be any anomalies in the measurements, nor should anything be heard that's less than desirable. Certainly, there are compromises that can't be avoided, but following these guidelines can result in taking a step toward the best possible outcome for your project.

Author's Note: Grateful acknowledgement is given to John Murray and Deward Timothy who first brought some of these concepts to my attention a long, long time ago.

Dan Alexander Audio: A Vintage Odyssey



Dan Alexander began buying and selling used guitars in 1967 at the age of 17. Until the late 1990s, he also built recording studios, sold hundreds of Neve mixing consoles, thousands of tube microphones, and all types of "old" recording equipment. His memories and notes are now compiled in a book. "I believe I coined the phrase 'vintage recording equipment,' an extension of my activities as a 'vintage' guitar dealer," he writes.

Review By
Scott Dorsey

For many years, old audio gear was considered junk of little value, much the way old video gear was. Back in the 1970s and early 1980s, nobody wanted old audio gear.

But there was a huge surge of interest in the late 1980s and early 1990s when "old junk" became valuable vintage equipment. That surge was promoted and supported by a few equipment vendors who specialized in reselling old equipment, and preeminent among those vendors was Dan Alexander.

Yes, there were plenty of other people out there—such as Fletcher from the Boston, MA-based audio dealer Mercenary Audio— who were also responsible for a big part of the vintage gear craze, but they didn't write books about it and Dan Alexander did.

Dan's book is a little bit fragmentary. It talks about Dan's experience buying and selling gear, and of expeditions to Europe to broadcast facilities where old Neumann microphones and Neve consoles could be had cheaply. It also talks a lot about the popular gear itself, exactly what it was and why people wanted (and still want) it. It also uses up a lot of pages on sales records about what Dan's company bought and sold at what time, which is probably of academic interest to a completist but isn't necessarily any more interesting than reading anyone else's sales records.

However, there is a lot of very cool stuff in this book. There are interesting stories, photos of gear, copies of old advertisements, and datasheets. Because it is so fragmentary, you can pick it up and start reading at any point and be amused. This makes it a great coffee table book, although it might not be the best coffee table book for your studio lobby because you might wind up with customers asking you why you don't have a Neumann M-50 like this book has in it.

Dan Alexander Audio: A Vintage Odyssey
By Dan Alexander
Rowman and Littlefield, Publishers
ISBN 978-1-5381-4201-1

Book Details Change in Attitudes

But what is most interesting to me is his discussion of the change in attitudes about vintage gear that took place at that time. Equipment that was going into the trash cans because nobody wanted that old crap was suddenly becoming valuable. People who had upgraded their U47 to nuvistor electronics were wishing they hadn't thrown out that awful VF14 tube because customers were interested in them.

The discussion of gear is by no means complete and centers mostly on the equipment that was most popular at the time. In many cases, it was popular in great part because of availability; every radio station in Europe seemed to have racks of Telefunken V72 mike preamps, and they were just flooding the market in the US. In other cases, it was specialized equipment that suddenly developed a demand.

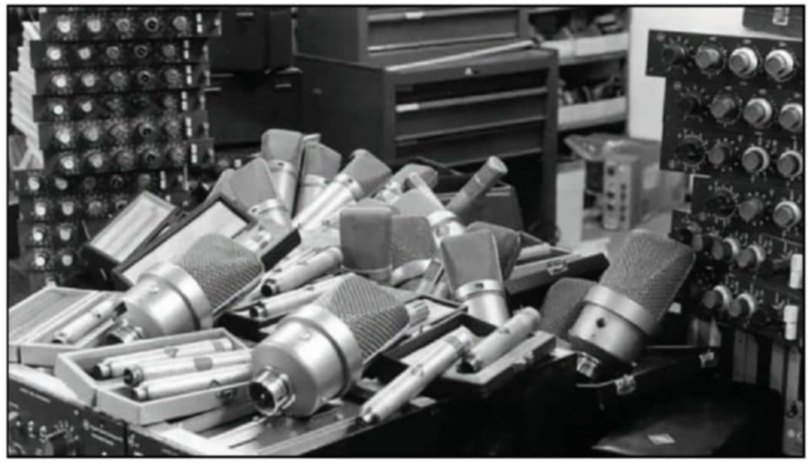
I lived through this era but I wasn't a part of it at all. Yeah, I bought some U47s for $150 each, but I bought them because they were the only decent microphones I could afford and I would have purchased a more modern small diaphragm mike if I'd had the money. I turned down a Fairchild 670 for $50 because it was too noisy and unstable compared with modern solid-state limiters whose thresholds never drifted. I watched the people snapping up this old equipment as the prices rose higher and higher with a combination of bemusement and horror, and I guess I still do have that attitude. This is part of what made Dan's book interesting to me, because he describes the motivations behind why people were buying and why he was promoting it.

A photo of an original Neve mixing console. DanAlexanderAudio.com is a treasure of audio industry resources, from specs to manuals and old brochures.

I can't unreservedly recommend this book. It's padded out with a lot of stuff that few people will find of interest, like the sales records and details of making albums for Dan's friend's band. I'd like to have seen more information about some of the more obscure but interesting audio manufacturers (e.g., MAP).

There are a few minor technical errors here and there but for the most part he avoids talking about technology and about what is inside the boxes to instead talk about who likes the boxes and where they came from. He doesn't mention the craze of repurposing every possible Neve module into a mike preamp, even modules that had nothing to do with the application. But that's not the point of this book.

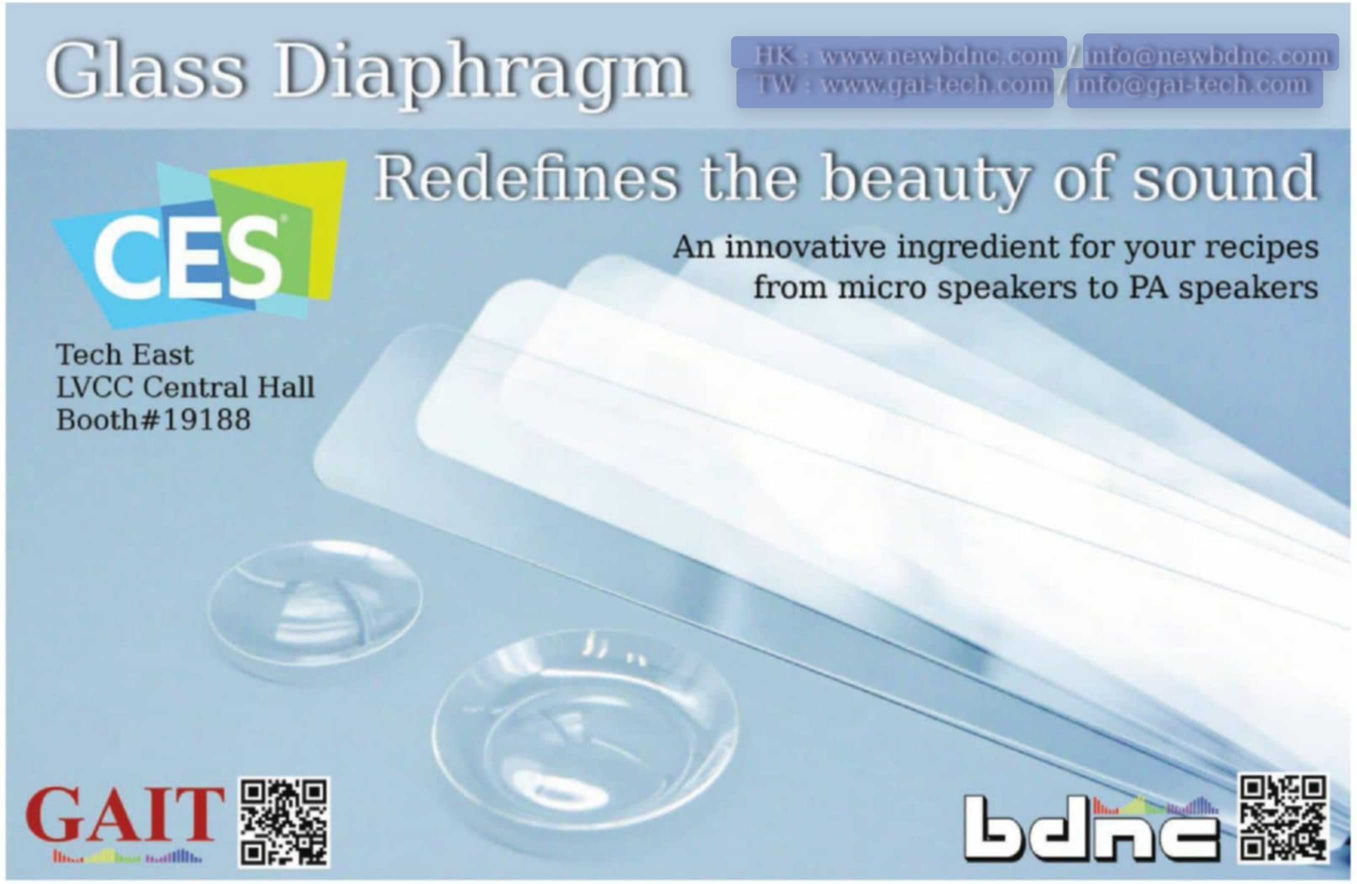
The point of this book is to talk about a time and a place and a movement in the audio world, and if you didn't live through it and want to know what it was like, Dan Alexander did live through it and he was central to it and he wrote about it for you.



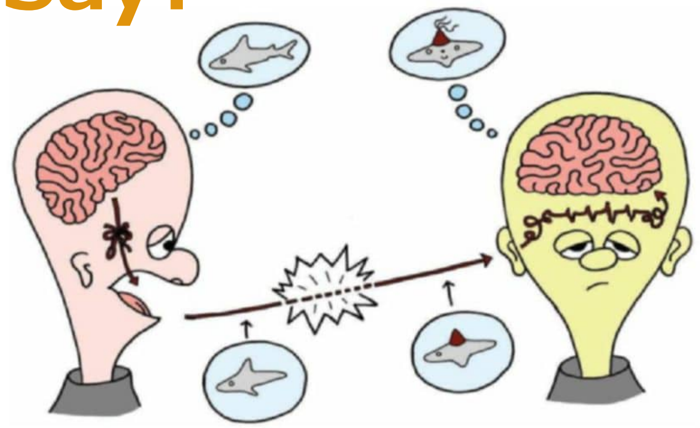
A pile of Neumann microphone and modules from Neve consoles at Dan Alexander's Hollywood store.

About the Author

Scott Dorsey has a degree in electrical engineering, during the pursuit of which he worked in the broadcast and recording industries. After several years working at a major studio, he took a job with a defense contractor. This left him time to do live concert recording for acoustical music and to design and build audio devices for personal use and on contract to several audio manufacturers and importers. Scott is a regular contributor to several audio magazines. He has been publishing equipment reviews and DIY projects since the mid-1980s. He is probably best known in the general audio community for his retrofit electronics designs in inexpensive Oktava, AKG, and Feilo microphones.



What Did They Say?



One frequent complaint that both acousticians and sound engineers hear is, "I can't hear what they're saying!" While this sounds simple, the complaint is fraught with complexities. In this article, our columnist shares some actual experiences with real and perceived speech intelligibility issues, before exploring measuring methods for quantifying it.

By
Richard Honeycutt

Really, You Can't Hear Them?

Often, "I can't hear..." really means "I can't understand..." The distinction is crucial. If the listener really can't hear the talker, increasing the sound level may be the solution; perhaps not, but it should be tried. If the problem is understanding speech, we call the issue poor Speech Intelligibility (SI). While proper sound level is important for good SI, it is no guarantee. As we will discuss later, poor SI is a many-faceted issue. If you're hunting rabbits, it does little good to be watching for groundhogs!

When I wrote telephone maintenance documents for Ma Bell, I found out that telephone servicemen responding to noise complaints were trained to ask, "What are you hearing: is it a crackling sound, a frying sound, a popping sound, a hum, or a buzz?" Giving the customer the proper vocabulary often helps immensely in solving a problem. The same is true with SI complaints.

We can ask, "Do you have this problem in only one place in the room, or everywhere? Have other people mentioned having the same problem? Is this a new problem, or has it been an issue for a while? Is the problem worse when the talker is male? ... female? Does it happen with all talkers, or just one or two? Is the problem related to the time of year?"

The answers to these questions may help identify the problem, as will become clearer later, when we discuss factors affecting SI.

Real-Life Stories

A colleague of mine provided acoustical design for an auditorium. He offered HVAC system noise analysis as one of his services, but the owner assured him that the architect's mechanical engineer was completely competent to design an appropriate system without the help of the acoustical consultant. About two weeks after the initial use of the venue, the consultant received a complaint from the owners that people couldn't understand the presenters. My colleague visited the auditorium and asked the building superintendent and his secretary to accompany him to the auditorium.

It being a chilly fall day, the HVAC system was in operation. They turned on the sound system and the superintendent was asked to read an article into the microphone while the secretary listened. She stated that she could not understand all that was being said, and in fact, neither could the consultant. Then while the superintendent continued reading, the consultant asked the secretary to turn off the HVAC system. Having done so, she exclaimed that the speech had become perfectly clear.

Unsurprisingly, when the consultant was later asked to analyze the HVAC system and his recommendations were implemented, the final cost of correcting the problems was higher than the cost of an initial acoustical analysis and implementation of the recommendations would have been.

A similar event involved another consultant colleague. In this case, HVAC analysis was included in his commission, but implementation of his recommendations was omitted as a "value engineering" step. The consultant only found this out after the owner complained of poor speech intelligibility and the consultant visited the site and examined the HVAC system to make sure his recommendations had been followed correctly. The retrofit to act on the recommendations involved stuttering and blushing on the part of the owner, followed by spending more money than it would have cost to do the job correctly in the first place.

In a more interesting and unique case, attendees complained of poor SI, but listening tests and measurements revealed no problem—"no problem" until the consultant attended a complete event on a hot day. Toward the end of the event, an ancillary HVAC system was activated, which was only used when temperatures in the room became too high for the main system to handle. Although this secondary system was shown in the drawings, and the consultant had recommended improvements, the improvements were not implemented since the ancillary system would not be used full time.

But unsatisfactory SI performance does not always involve HVAC systems. A sound-system designer I knew responded to a request to investigate poor SI in a recently completed venue, which he had not designed. He asked the owner and his secretary to demonstrate the problem. The sound system was powered up, the owner began speaking, and the consultant and secretary agreed that the SI was not acceptable. Then the consultant powered down the sound system, and suddenly, speech was quite understandable. A sound-system redesign and replacement completed this success story.

In another instance, some congregants at a new church sanctuary complained of poor SI, even though they were sitting close to the speakers. An investigation showed that the speakers in question were artifacts from an old system that had not been removed when the new system was installed. Removing the old speakers eliminated the psychoacoustic problem that had caused the listeners to think SI was poor, and the complaints ceased.

Two of my own clients had similar experiences around the same time. In the first, the SI complaints began soon after the arrival of a new minister who was very soft-spoken. Some improvement was noticed after heroic efforts were made to improve the already-good sound system. I concluded that nothing further could be done, since the sanctuary was designed to support pipe organ and choral singing, thus having an intentionally long reverberation time (RT). But a few months later, when the minister was reassigned to another church, SI complaints ended.

Another church in the same city was planning a renovation, and I was invited to a meeting involving the hearing-challenged chairman of the renovation committee and the pastor. The pastor had a deep, rich voice, but as sometimes occurs, with age his enunciation of important consonants had deteriorated. At one point when we were discussing objectives for the new sound system, the pastor said to the committee chair, "If some hard-of-hearing people would get their hearing aids adjusted, we wouldn't have a problem." To which the committee chair replied, "If some people would take the marbles out of their mouths when they preach, we wouldn't have a problem." Both men were partially right: It can be impossible to provide good enough SI to please hearing-challenged listeners, and there are limits to the acoustical and technological fixes to overcome poor enunciation and vocal projection. Earset mics can help solve such problems, but some talkers refuse to use them.

Another client of mine asked me to investigate SI complaints in his church sanctuary. He said that in addition to intelligibility issues, the system sometimes "reverberated." I found that the SI issues resulted mainly from the use of pre-teen or elderly lay readers who did not know how to project their voices. The "reverberation" came from the sound operator pushing the mic levels up to the brink of feedback squeal (no reverberation was involved) in an attempt to overcome the weak voice level at the mic.

In one other example, a colleague recounted the story of a train station whose customers complained of poor SI especially in the evening. Investigation revealed that the announcer on duty in the evening was so obese that he could not sit closer than 6' to the microphone. In this case, a headset mic solved the problem.

In Conclusion

Sometimes an SI problem is caused by acoustical issues. The Acoustical Society of America (ASA) released a proposed Classroom Acoustics Standard, which has been codified as ANSI Standard S12.60. This standard specifies maximum background noise levels and RTs for school classrooms and other spaces. I worked on a project in which a particular county upgraded a group of schools to meet the acoustical standards. After renovation of the first school was complete, a faculty member complained to the administration that the reverberation seemed a bit high in some of the rooms. Measurements revealed that she was correct.

Checking upon the implementation of our recommendations, we found that certain approaches we had recommended to control RT in a number of the rooms had been vetoed by the school board, over the architect's objections. We had not been informed. Physics, however, is not subject to budgetary or political considerations, and physics prevailed. In this case, the only beneficiary was the General Contractor, who was paid to retrofit the spaces according to our recommendations, although—truth be told—he would have preferred to avoid the refit and be able to concentrate on new construction!

In the next Sound Control article, we'll look at design elements for good SI and review the development of standardized SI measurements.

About the Author

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

Olive Pro Hearing and Bluetooth Earbuds

The Challenge of Hearing Enhancement

By
Brent Butterworth

Photo 1: Following its first presentation at CES 2020, Olive Union officially made its 2-in-1 true wireless earbuds and smart (FDA-Registered class II) hearing aids available for purchase in September 2021.

The Olive Pro personal sound amplification product got mixed reviews, and the company has temporarily paused its sales efforts in the US. Brent Butterworth tested the product, then talked with the company to find out what happened and their plans for the future.

Hearing aids and personal sound amplification products (PSAPs) seem to be the most talked-about category in audio right now—yet it also seems to be the most tumultuous category in audio. When Congress passed the FDA Reauthorization Act of 2017, it mandated the creation of a new class of over-the-counter (OTC) hearing aids that consumers could purchase and configure on their own, without help from an audiologist. But the US Food and Drug Administration, slowed by the COVID-19 pandemic, did not release proposed guidelines for the design of OTC hearing aids until October 2021, and it will take several more months, at least, for the agency to collect feedback from the public and the industry to incorporate the feedback into a final set of guidelines.

Despite a lack of federal governmental guidance and a confusing tangle of state regulations, several companies have, in this five-year interim, created OTC hearing assistance products that, in essence, repurpose the now-familiar true wireless stereo (TWS) earphone concept for hearing augmentation. After all, a hearing aid is just a microphone, an amplifier, a driver, and some audio signal processing—components found in every set of TWS earphones, which are now a commodity item selling for as little as $20.

One of these companies is Olive Union, which launched its second-generation PSAP, the $299 Olive Pro, in December 2020. The Olive Pro is, in essence, a set of TWS earphones pressed into duty as PSAPs (**Photo 1**). Like all TWS earphones, they can be used for music listening and phone calls, and they also incorporate active noise cancelling (ANC). But they add an app that offers the user a hearing test, then provides three different hearing enhancement modes (Boost, Clear, and Quiet), which can be set for enhancement levels from 0 to 10.

The effect of these modes can be individually fine-tuned for each ear using a 10-band EQ, and the playback response for music listening can be programmed through a separate EQ. Although the Olive Pro can't be programmed by an audiologist, it's possible that a sophisticated user could take the results of an audiogram and use those with the EQ to get a more accurate compensation for their hearing loss.

There aren't a lot of publications reviewing hearing aid products, but still, the Olive Pro got mixed reviews, mostly complaining about the sound quality with music playback. We'd already started digging into an extensive test of the product when, in early fall 2021, the company announced it was pausing sales in the US market. A couple of

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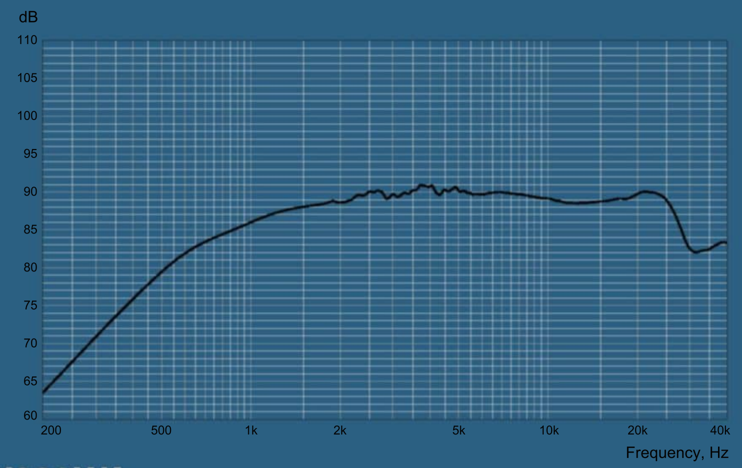

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weeks later, we interviewed some of the company's executives via Zoom to find out why they hit pause, and what the future of the company looks like.

This article will share our findings about the current version of the Olive Pro, along with the company's initial vision for the product, its plans for future operations in the US, and its analysis of the current state of the hearing aid and PSAP markets.

The PSAP Challenge

The idea behind Olive Union's hearing-assistance products is basically the same as that of most competing products, but Olive Union does have its own twist on it. As Chief Technology Officer Yoon Soo framed the challenge, "Eighty percent of the people who could benefit from hearing aids have mild to moderate hearing loss, but the penetration rate—the number of those people actually using hearing aids—is only around 15%. We wondered where that discrepancy is coming from, and our research showed that people who could benefit from hearing aids most often don't wear them because of usability issues, like getting them to fit right, and having them tuned to deliver a clear benefit. So we decided to make a hearing aid that looks like a Bluetooth earbud, is easy to use like AirPods, and could be used in one ear or two."

Chief Operating Officer Teo Son added, "We want to make an appealing product that they'll want to wear, instead of one they'll be reluctant to wear. We think the Olive Pro is a perfect example of the hearing aids that the U.S. Food and Drug Administration [FDA] wants to make available over-the-counter."

In my discussion with Olive Union, it became clear that the company's unique selling prospect—in what has already become a crowded field—is to offer a "just right" amount of user options, enough that users can customize the product's response to their needs and tastes, but not enough that they become intimidated or need to understand the technical aspects of the product.

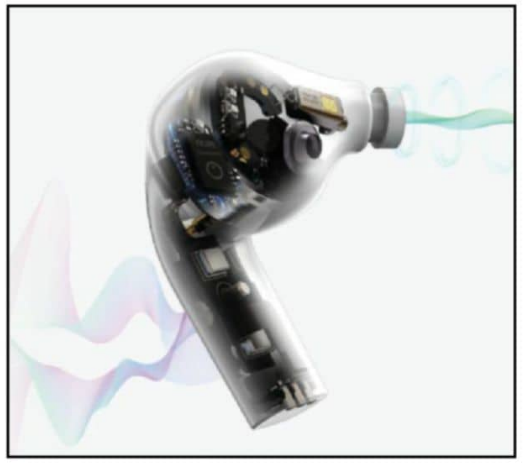


Photo 2: The Olive Pro earbuds connect via Bluetooth 5.1 with SBC/AAC support and feature built-in processing power to handle active noise cancellation plus hearing assistance and customizable passthrough audio, 10-band EQ for assistance or music playback, and multi-band wide dynamic range compression.

As Soo put it, "High-end hearing aids have different frequency bands that can be tuned, and they place a lot of emphasis on custom tuning by an audiologist. With personal sound amplification products such as the Nuheara [reviewed in the July 2021 *audioXpress*], you get just a few modes, like drive, outdoor, etc., and you can't really go in and tune the frequency bands and the gain level. We wanted to include as many custom tuning features as possible, not just gain but also attack and release for the compression. We wanted to offer them those adjustments in a way that they're easy to use and the customer doesn't have to understand the technical aspects of it."

Some of these features are already built into the app, such as the ability to equalize the sound differently for music playback than for hearing assistance, using a graphic equalizer. According to the US team lead Nicholas Dahl, "We are working on an update for late 2021 that will give you more controls, such as custom-tuning the gain level. We plan to use more parameters that a layman can understand. They don't understand attack and release, but they do understand things like noise, treble, bass, and speech clarity. So we give them controls for these parameters, but underneath all that we change the attack, release, and ratio of the compressor."

During our talk, the Olive Union execs stressed the need for the features included on a PSAP to deliver an obvious, easily demonstrable benefit, so that users can be confident it's working and more likely to appreciate what the device can do for them. Son noted this principle when I asked him why the Olive Pro doesn't include a directional focus feature, which many current PSAPs offer. "We have the technology to include directional pickup, but like I saw you found when you tested the Nuheara IQbuds2 Max for *audioXpress*, most of those only give you a few decibels of boost on sounds coming from the front," he said. "We have been able to push this to around 7dB to 8dB, but our marketing team was expecting more like 12dB, so they weren't satisfied with the results we got and they decided not to go ahead with it for now."

What the company discovered, though, is that the PSAP market introduces problems engineering and product planning can't solve. "I think the audio engineering wasn't as big a challenge as the ergonomics. There are a lot of expectations when people buy hearing aids," Soo said.

The Problems

PSAPs have been around for several years, incorporating increasingly advanced technology while becoming sleeker and stealthier, but none of them appears to have been a major success. Indeed, while we were preparing this article, the company decided, on a temporary basis, to stop selling its product in the North American market. I was curious to find out why, considering that some of the complaints about performance that I'd read in early reviews had already been addressed in updates to the app.

Dahl said that the company's actions weren't due to engineering difficulties or product flaws. "We decided to pause sales in North America because even though we've done a lot to make this product user-friendly and easy to set up, we found many customers still need help with the product," he said. "So we're building up our customer support operations now, with a 24-hour call center for US customers, and I think we will be able to relaunch the product toward the end of 2021. Having the call centers will also give us a lot more feedback from customers, which will help us improve the product."

According to Olive Union execs—and to every one of the many audiologists I've interviewed in the past two years—the biggest problem with hearing aids and PSAPs is that they initially feel and sound strange, because in the process of trying to correct for flaws in the listener's hearing, they present sound in a very different way than that to which the user has become accustomed.

"Glasses work instantly—you can see the benefit right away," Soo pointed out. "But hearing aids aren't effective right away—you have to get accustomed to them, and they may need some further tuning beyond what's achieved automatically through the hearing test in the app. We—and the rest of the industry, too—need to do a better job of explaining to customers what the hearing aid process is going to be like."

The Product—and Specifically the App

I got the chance to spend a couple of months using the Olive Pro PSAPs, and to run an extensive set of measurements to gauge the product's performance and see where there's room for improvement.

As with most PSAPs, the Olive Pro's app is probably more important and interesting than the hardware (**Photo 2**). After you've installed the My Olive app, a screen appears that shows the average level of external sound coming through to your eardrums, and the average level of your music listening—both good things to know if you're concerned about your hearing. It even shows you a graph that tracks your listening levels day-to-day (**Figure 1**). In my opinion, every app included with every set of Bluetooth headphones should include this feature, because the volume setting on the device is known, the gain of the amp and the sensitivity of the driver are known, and the average level of the content can easily be gauged, so the SPL can be known with reasonable and useful accuracy.

A screen below these meters prompts you to optimize the Olive Pro's performance using a five-minute hearing test to create what the company calls a HearID. Like most hearing tests included with PSAPs, this one plays a certain frequency in one of your ears at a moderate level, then backs off the level in steps of several decibels, and asks you to press an onscreen button when you hear the tone again. Unfortunately, the time between the tones is always the same, and I found that I sometimes reflexively anticipated the next tone, pushing the button even

Figure 1: Olive app readout of average listening level over the course of six days

if I wasn't sure I heard the tone. This is a problem that medical hearing tests take care to avoid.

Regardless, the test seemed to profile my hearing fairly accurately. Once the test is done, it shows a graph (**Figure 2**) similar to what you'd get from an audiologist, detailing the sensitivity of your hearing in each ear across most of the audio range. Unfortunately, this graph is not labeled with frequency or decibel information, so its representation is unnecessarily abstract—but it does look much like the results of my most recent hearing evaluation, conducted in the process of my February 2021 *audioXpress* review of the Widex Moment hearing aids.

As noted previously, the app gives you extensive control over the hear-through (hearing enhancement) function, noise cancelling, and music playback (**Figure 3**). When you click on one of the hear-through modes, a frequency response plot shows a combination of the hearing correction it has applied (based on your hearing test results) in each channel, the transfer function of the listening mode you've chosen, and the hearing EQ settings you've added, so you can always see exactly what the Olive Pro is doing. At least for users who have some basic comprehension of audio, this should be a big help in allowing them to tune the product to suit their needs.

I found all of the controls and functions intuitive to use. The app has an integral user guide to guide you through the operations, but I doubt you'll need it.

Listening

Fortunately, my hearing loss is typical for my age (59) and gender, so I don't really need hearing correction, but having evaluated and measured one hearing aid and several PSAPs, I have a decent understanding of what these products are supposed to do.

With three different hear-through modes, separate EQ for each ear, and adjustable

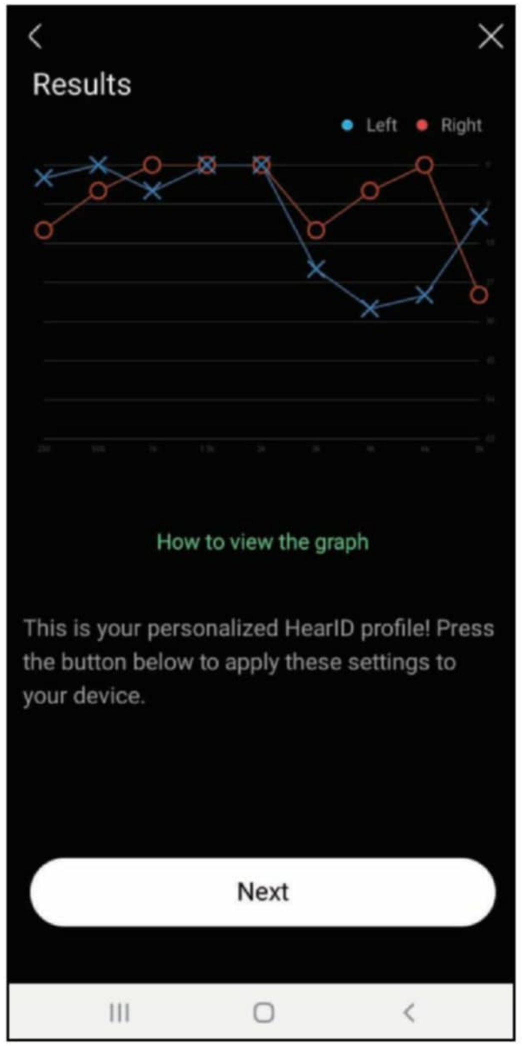


Figure 2: Results of the author's hearing test—the HearID—shown on the Olive app

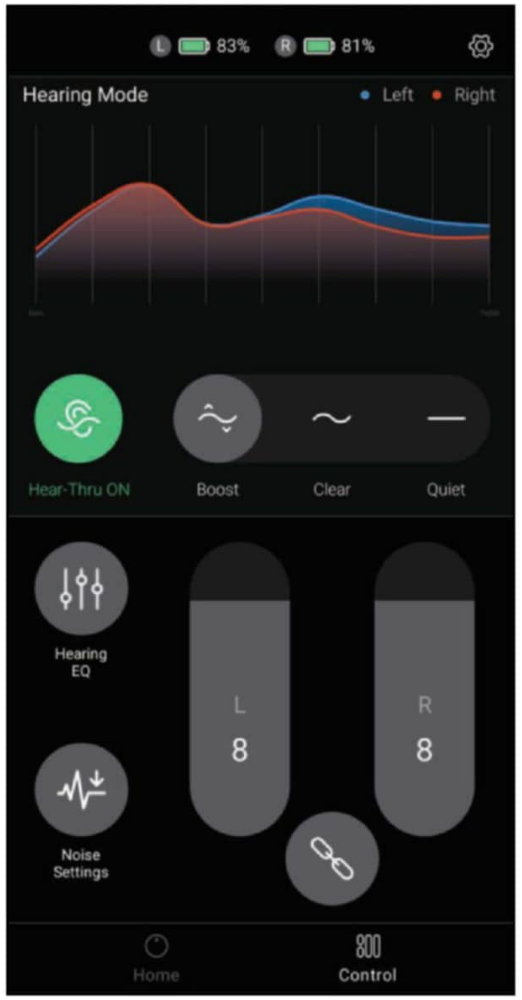


Figure 3: The main control screen of the Olive app

compensation level, it's tough to characterize what the Olive Pro does, and in some ways, it's tough to criticize—because if there's something you don't like about what it's doing, there's a good chance you can correct it, at least to some extent, through the controls. (Although there's one exception, which we'll address shortly.)

Of the three modes—Boost, Clear, and Quiet—Boost had the most dramatic and useful effect, thanks to a large peak in the lower treble. However, it adds a lower midrange boost (perhaps intended to aid music listening) that adds a persistent "whooshing" sound. I was able to get rid of this effect by turning down the bass and lower mids. The Clear mode simply elevated the treble, while the Quiet mode only boosted the lower treble. I spent a lot of time in a lot of different settings with the Olive Pro, listening for how it enhanced voices, and I found that none of the presets achieved the clarity enhancement I've gotten with the best PSAPs—unless I applied my own EQ.

After some experimentation, I got it to the point where I heard a substantial increase in voice clarity. For example, I was watching a kids' football coach at a distance of perhaps 25 meters, and without the Olive Pros, I could just make out the calls he was shouting. With my custom EQ setting, and the enhancement cranked up to 8, 9, or 10, it was easy to hear and understand the calls—the Olive Pro felt almost like a "sonic telescope," even though it lacks a directional pickup function.

Music listening through the Olive Pro was OK. To me, the sound was comparable to that of typical inexpensive wired earphones—listenable, but lacking in detail and depth. I was able to improve on this with the music EQ in the Settings menu—using the Bass Boost mode with a little extra lower treble added—but I never got sound that really impressed me.

Soo told me that the company didn't tune the

Photo 3: The Olive Pro earbuds are shown under test using the GRAS RA0402 high-resolution ear simulator/coupler.

Olive Pro to any specific response, such as the Harman curve, and that the engineers sought "a natural balance of bass to treble." He pointed out that the app gives listeners the power to tune the sound to their liking—and maybe for PSAPs, where the listeners will have a widely varying degree of hearing loss, likely including "noise notches" in the upper midrange and lower treble and different levels of loss in each ear, tuning for a certain ideal response doesn't make much sense.

The thing that really bothered me about the Olive Pro's performance is its latency. All hearing aids and PSAPs have some latency, which if it's long enough, can create two problems. First is echo, as you hear the direct sound leaking through the ear tips or bone conduction, followed by the processed sound, sometimes many milliseconds later. Second is the weirdness of seeing your hand click, say, a knife against a cutting board, and having what arrives at your ears lag noticeably behind what arrives at your eyes—much like experiencing a lip-sync problem when watching a digital TV broadcast.

Although the Olive Pro's measured latency varied greatly from one trial to the next, the latency was always high enough to create the echo effect. It reminded me of some old rock recordings, such as the famous drum sound from Led Zeppelin's "When the Levee Breaks," which thickened the beats with slapback echo from a Binson Echorec magnetic echo machine. I could hear a little bit of this effect when I tested the Nuheara IQbuds2 Max, but it was never enough to bother me.

I tried the noise cancelling function at home, with pink noise playing through speakers, and on a jet airplane trip from Los Angeles, CA, to Seattle, WA. I heard very little noise reduction when the cancellation was activated, and only a slight difference between the different noise cancelling modes.

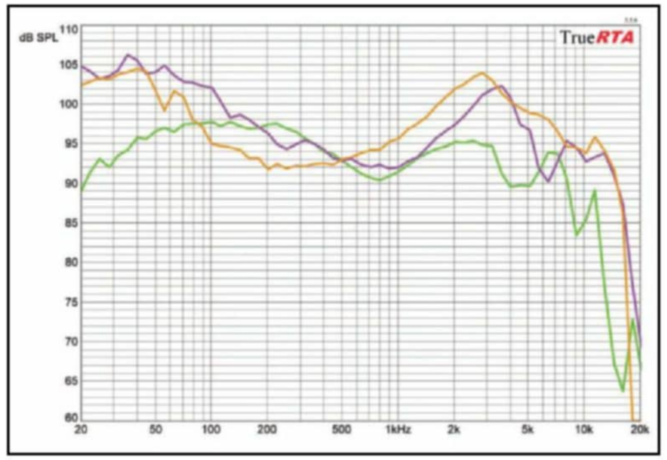


Figure 4: Frequency response of Olive Pro without hearing compensation (green), Status Audio Between Pro (purple), and JBL Club Pro+ (orange)

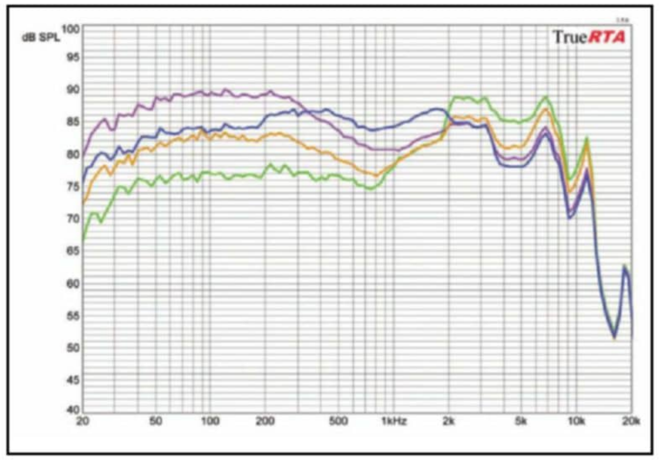


Figure 5: Function of the Basic EQ settings, in off (green), Bass Boost (purple), Treble Boost (orange), and Vocal Boost (blue)

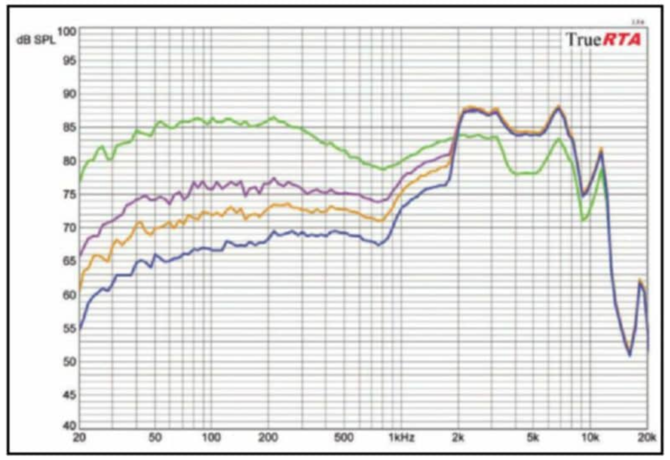


Figure 6: Function of the Special EQ settings, in off (green), Level 1 (purple), Level 2 (orange), and Level 3 (blue)

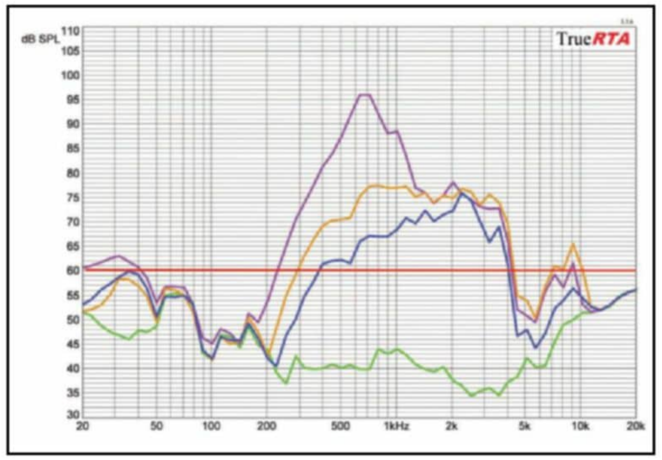


Figure 7: Output of the Olive Pro with a 60dB external noise level, with hear-through off (green), and in Boost (purple), Clear (orange), and Quiet (blue) models

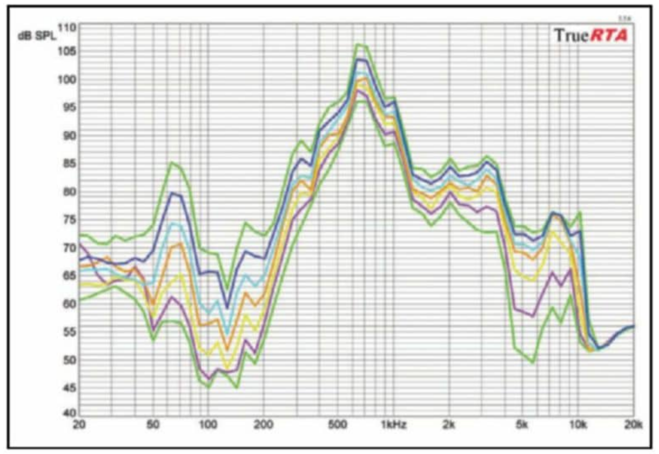


Figure 8: Output of the Olive Pro in Boost mode starting at a 60dB SPL external noise level (bottom green line) and going up in 5dB increments to a maximum of 90dB

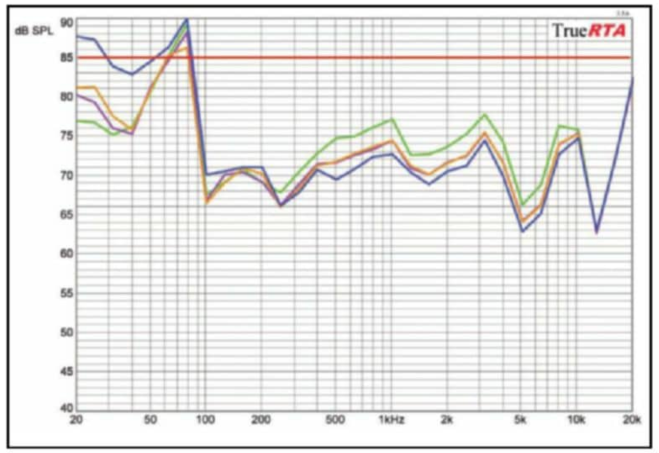


Figure 9: Isolation of the Olive Pro, at a baseline noise level of 85dB SPL, with noise canceling off (green), and at Level 1 (purple), Level 2 (orange), and Level 3 (blue)

Measurements

I performed measurements using a GRAS RA0402 high-resolution ear simulator/coupler (**Photo 3**). For latency measurements, I used an Audiomatica Clio 12 QC audio analyzer, feeding signals into the Olive Pro via a Reiyin WT-04 USB Bluetooth adapter. For all other measurements, I used the RA0402 with an M-Audio Mobile Pre USB interface running TrueRTA software. (Incidentally, Olive Union uses a standard 711 coupler, which is like the RA0402 but has a much stronger ear canal resonance—typically around 13kHz—but it is dependent on the device under test.)

The latency of the Olive Pro jumped around sporadically—measurement-to-measurement variation on the order of 10ms to 30ms is common with true wireless earphones, but with the Olive Pro, it was sometimes as low as 10ms, but occasionally ran longer than 300ms. Thus, I was unable to set the Clio's gating to compensate for the latency. For this reason, I had to use pink noise with the TrueRTA spectrum analyzer to do the frequency response measurements. I cannot recall seeing such large variations in latency with any Bluetooth headphones or earphones I've measured, although based on my listening, I have to think the 300ms figure is more representative of this product's performance.

The Olive Pro's "native" frequency response—with no hearing correction applied—for music playback is shown along with the responses for the JBL Club Pro+ and Status Audio Between Pro true wireless earphones (**Figure 4**). The Olive Pro's frequency response shows flatter output than the others, but with headphones, a flat response suggests a headphone will sound midrange-heavy. It also shows more output between 100Hz and 300Hz than the others, and much less output in the treble and upper midrange, above about 1kHz. This doesn't suggest particularly good sound, but of course, the app can compensate with the 10-band EQ, the Special EQ settings to compensate for hearing loss (**Figure 5**), and the three different Basic EQ music listening presets (**Figure 6**).

The transfer function of the hear-through modes (**Figure 7**), measured here at maximum boost with pink noise at a base SPL of 60dB, is fairly typical for a PSAP, with a boost focused between 300Hz and 4kHz. Maximum boost occurs in Boost mode, with peak amplification of 36dB centered at 650Hz.

Compression—raising the level of critical frequencies while applying a limiter to them, so they're always heavily boosted—is a standard feature of hearing aids. To profile the characteristics the Olive Pro's compression function (**Figure 8**), I started in the Boost mode at a baseline uncorrelated external noise level of 60dB SPL, and raised the noise level in the room by 5dB increments to a maximum of 90dB SPL. As you can see, compression is highest between about 300Hz and 2kHz, where the 30dB signal level increase yields an output increase of only 10dB, for a compression ratio of roughly 3:1. There's little or no compression in the bass, and compression in the treble didn't kick in until the level got louder than 70dB.

To measure isolation—and in particular, the effect of the noise cancelling—I played uncorrelated pink noise at 85dB SPL through a six-speaker system and used TrueRTA to see how much sound leaked through at each frequency (**Figure 9**). There was little measured improvement with noise cancelling activated.

Resources

B. Butterworth, "Nuheara IQbuds2 MAX Hearing-Assist Earphones," *audioXpress*, July 2021, https://audioxpress.com/article/fresh-from-the-bench-nuheara-iqbuds2-max-hearing-assist-earphones

B. Butterworth, "Widex Moment mRIC RD Hearing Aid," *audioXpress*, February 2021, https://audioxpress.com/article/fresh-from-the-bench-widex-moment-mric-rd-hearing-aid

Olive Union, www.oliveunion.com

Conclusion

Over the course of the last two years, I've spoken with numerous audiologists, hearing experts, and manufacturers of hearing aids and PSAPs. Yet, I still feel we're all groping around in the dark. There's little consumer research we can tap, and very few in-depth reviews we can consult or use as a paradigm. And while there are predictable patterns to hearing loss, everyone's hearing loss is different, and as Olive Union's execs noted, emotional issues and often-unrealistic user expectations make it hard to speculate about what hearing aid or PSAP will work well for which person (**Photo 4**).

That said, I think the Olive Pro has enough power and flexibility to help many people who suffer hearing loss—but not enough hearing loss that they want to spend thousands of dollars on audiologist-fitted hearing aids. However, I expect the latency in the product's processing will bother many users, and I wonder how many of the people in the demographic most likely to experience hearing loss will have the knowledge and the patience to get the Olive Pro dialed in for their ears. But of course, this product, like all app-driven PSAPs, is a work in progress, and Olive Union anticipates that it may make some changes to the app—and thus, the Olive Pro's functionality—once the FDA has settled on a firm set of guidelines.

Photo 4: The Olive Pro's modern design was intentionally conceived to look like futuristic Bluetooth earbuds to combat the social and economic barriers of hearing loss.

About the Author

Brent Butterworth has worked as an audio journalist and technical consultant since 1989. He currently writes for *Wirecutter*, *SoundStage*, and *JazzTimes*, and has published speaker, headphone, and amplifier measurements in *audioXpress*, *Sound & Vision*, *Home Theater*, *Home Theater Review*, and many other publications. He has also served as a consultant on measurement, design, and tuning of audio products, with a special focus on active speakers and DSP tuning.

Designed for Growth

Enabling Product Differentiation for TWS Development

By
Raj Senguttuvan
(Knowles Corp.)

With the availability of an AI-enabled True Wireless Stereo (TWS) development platform, Knowles is looking to accelerate the product development process, from entry-level to premium applications. The new Knowles TWS development kit shortens the design cycle for OEMs, enabling advanced, differentiated products.

Knowles announced the availability of a new development kit for true wireless stereo (TWS) earbuds, centered on the capabilities of its AISonic family of audio edge-AI processors and key audio processing partnerships. This article details the opportunities for TWS development and helping manufacturers to differentiate their newest designs.

The market for true wireless stereo (TWS) and high-quality audio is growing fast. Driven by strong demand across all price segments and listeners—from mainstream consumers on a budget to audio pros—the pressure for premium brands to add emerging, advanced features to their TWS devices is a constant. Canalys predicts that 2 billion TWS earbuds will be sold by 2024, a projection rooted in a remarkable year-over-year growth of 90% between 2019 and 2020.

New use cases for TWS headphones (e.g., voice calling with advanced noise suppression features, HD audio streaming, adaptive active noise cancellation, and hearing or conversation enhancement) will all continue to accelerate the demand from consumers. As OEMs race to develop their own versions of in-demand TWS designs, product differentiation is the key to ensuring their piece of the market share.

To empower OEMs, the creators of these sought-after devices, Knowles recently created a configurable powerful TWS development kit that provides them access to fully operational TWS design. Enhanced with the most advanced features and functionality available today, the kit comes with pre-tuned earbud configurations equipped with Knowles' SiSonic MEMS microphones, voice vibration sensors for high-quality audio capture, and a hybrid or full range balanced armature speaker driver assemblies that can support high-resolution audio specifications.

Here, we'll dive into some of the specifications of the kit—the technical must-haves for OEMs continuing to ride the wave of TWS growth. This article will also explore how an advanced TWS development kit loaded with computational horsepower and features can drive the path to success in the TWS market.

The Competitive Edge: What's Under the Hood

The processing platform at the core of the Knowles TWS Development Kit is based on the IA8201 AISonic audio edge processor, Sony Semiconductor Corp.'s CXD3781 codec, and Qualcomm's 5141 premium Bluetooth SoC chipset.

The IA8201 processor provides audio fidelity and machine-learning optimization, allowing manufacturers to deliver a competitive edge for high-quality audio communication devices and other advanced AI features. It brings the advantage of high-compute efficiency and low-power operation for voice activation, hands-free control, and contextual audio processing. It also features a rich set of audio and general-purpose high-speed interfaces to enable flexible performance with digital microphones and other sensors. And crucially, the open DSP available on the IA8201 audio edge processor provides a rich, scalable platform for customer or third-party solutions.

The IA8201 is also a great choice for TWS manufacturers looking to quickly add third-party algorithms and AI-based voice or audio features with a short time-to-market requirement. The Knowles TWS kit comes with several third-party algorithms pre-integrated and optimized on IA8201 that deliver high-performance and differentiation.

What Listeners Want to Hear: Driver Optimization for ANC, Output, and More

The TWS marketing and consumer preferences are trending toward premium audio. More consumers are demanding hardware that offers studio-quality audio output for enjoying their favorite music. Since Bluetooth and Cloud connectivity has made massive improvements since the MP3 era, HD or lossless quality audio can now be directly streamed to TWS devices. In addition, advancements in digital signal processing (DSP) and driver technologies means TWS devices can support high-fidelity music while also featuring state-of-the-art ANC and transparency features. The combination of advanced features offers a fresh listening experience, even for seasoned enthusiasts, making it necessary for OEMs to embrace this philosophy in the product development process.

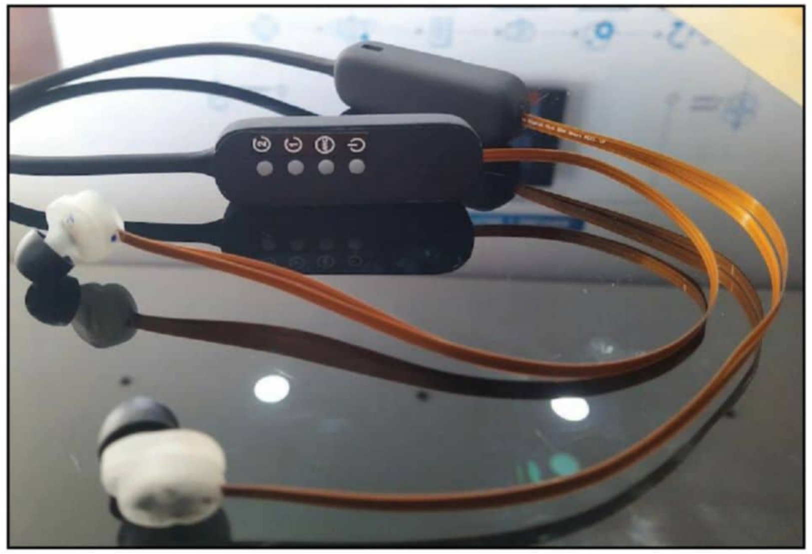
Balanced armatures (BAs), developed originally for hearing aids and critical listening applications, offer improved fidelity and detail that remain unmatched by traditional dynamic drivers. Where dynamic drivers operate with a moving coil, BA drivers use a stationary coil that vibrates a reed balanced between two magnets inside a small enclosure. The lighter mass of the moving parts results in exceptional clarity, especially in the treble. Its lightweight and compact footprint also saves space for additional components, such as additional drivers for hybrid applications or for bigger batteries.

BA drivers may also be combined with traditional dynamic drivers in a hybrid system. In these systems, the dynamic speaker handles the lower frequencies, while the BA driver(s) reproduce the critical highs. The Knowles TWS Reference Design Kit offers manufacturers two versions of earbuds to embrace this key benefit—one with a full-range BA and the other with a hybrid driver, with a BA tweeter and dynamic driver woofer. Both earbuds have three digital microphones with high signal-to-noise ratio (SNR), low distortion, low latency, and a flat frequency response to ensure premium performance.

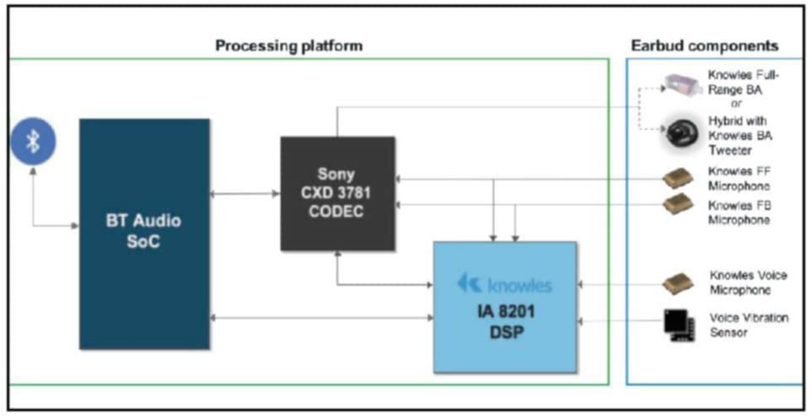
Hearing Augmentation Gets a Boost

Recent industry research from FutureSource confirms that conversation enhancement through hearing augmentation is highly sought-after by 61% of TWS earbud users. Paralleling the hearing aid industry to improve sound perception, hearing enhancement enables users to enjoy more vivid conversation, used with or without ANC depending on the environment. When applied to TWS use cases, a booster algorithm removes disruptive background noise including music, reverb, and surrounding conversation for clean sound processing, reducing the listening effort for users.

The Knowles TWS Development Kit includes AITransparency+, a feature from Knowles partner Chatable—the first AI specifically designed to exploit the next-generation AI processing capabilities of the IA8201 AISonic processor for "Conversation Enhancement." AITransparency+ provides "Conversation Enhancement" and features advanced, on-chip, proprietary deep neural network architectures performing more than 100 million AI calculations per second to enable selective acoustic enhancement of conversational speech signals without perceptible latency.



With this new development solution from Knowles, OEMs gain access to a fully operational TWS development kit, which includes pre-tuned and pre-configured earbuds designed by Knowles and paired with a powerful Bluetooth-enabled processing platform (Qualcomm). The earbuds come designed with Knowles SiSonic MEMS Microphone Arrays, voice vibration sensors, and a choice of premium speaker driver assemblies with Knowles' Balanced Armatures to drive the development of advanced TWS features, while reducing time-to-market and lowering the high costs associated with ground-up development.



The Knowles IA8201 AISonic audio edge processor is optimized for advanced voice and audio processing. This development platform is flexible, allowing manufacturers to test their own configurations with detachable earbuds, integrate advanced features, and tune for performance and power consumption.

Getting by with a Little Help from Friends: Personalization for TWS Development

Designing an advanced TWS kit takes a broad range of technologies from multiple domains. This is why the Knowles development kit has been developed in partnership with several algorithm vendors to bring a breadth of advanced features and shorten the design and market cycle for OEMs that want to deliver emerging use cases. With existing integrations from Sony, Chatable, Alango, and Sensory, the Knowles TWS kit already has the most advanced algorithms for TWS devices today.

Combined with proprietary technology, Sony Semiconductor's CXD 3781 codec is included in the design to provide integrated hybrid noise cancelling and transparency. This works in tandem with a QCC 5141 Bluetooth Audio SoC which, along with premium BAs, deliver high-resolution audio.

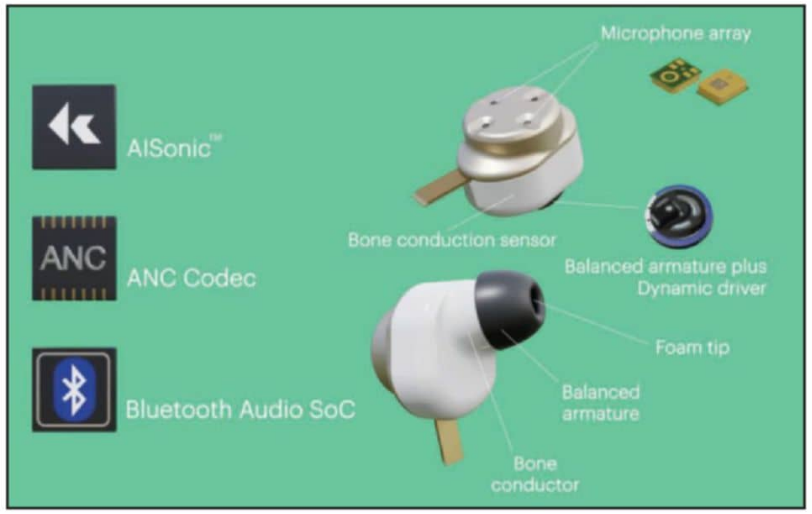
To enhance voice quality and intelligibility, the kit is also supported by an advanced voice communication package from Alango Technologies, which offers multi-microphone beamforming, and its OnlyVoice technology, which intelligently combines external beamforming with internal sensor-based voice processing ported to the IA8201 processor.

With low power consumption and memory requirements, these highly optimized solutions enable premium voice pickup for TWS devices. By combining Sensory's wake word technology with the Knowles' IA8201 audio edge processor, manufacturers can achieve high-performance, embedded voice solutions.

The inclusion of technologies from these partners in the Knowles TWS kit enables OEMs to benefit from established technologies, while also including their own third-party or proprietary features in their designs. The integration of the most in-demand features will allow for a competitive edge, with the opportunity to continue to build and modify based on the manufacturer's expertise.

Conclusion

The TWS development kit from Knowles builds bridges for TWS developers: It grants access to the most in-demand features for forward innovation, supported by premium Knowles components and state-of-the-art partner technologies. It also allows more partners to join in on the growth with an accessible option for product development. Offering the foundation for TWS technology allows us to reach more OEMs, diversifying the product mix available to consumers across the full spectrum of audio consumption.



The Knowles Development Kit for True Wireless Stereo (TWS) earbuds is centered on the company's microphone arrays, balanced armatures, bone conduction sensors , plus the capabilities of its AISonic family of edge-AI processors to run voice and audio processing algorithms, paired with a codec that provides integrated hybrid noise cancelling, AFE, ADC, DAC, and filters, and a premium Bluetooth Audio System-on-Chip (SoC).

The TWS market is not just expected to grow; it's already soaring. It's up to manufacturers to meet the fast-raising bar, and with the right tools, it's getting a whole lot easier. The features offered by the TWS development kit enable earphone designers to meet popular use cases (e.g., premium sound, ANC, conversation enhancement, and more). The open, configurable nature of this kit supports existing and emerging TWS premium features and next-generation applications, leveraging best-in-class audio and hearing technologies with flexibility for future innovation.

www.knowles.com

Resources

"Music, Work, Gaming, and Health—The Growing Role of Headphones Post-Pandemic," FutureSource, www.futuresource-consulting.com/insights/music-work-gaming-and-health-the-growing-role-of-headphones-post-pandemic

About the Author

Raj Senguttuvan is Director of Strategic Marketing at Knowles Corp., where he leads Knowles Audio Solutions strategy, and the development of advanced solutions for emerging Consumer and IoT applications. He holds a PhD in Electrical Engineering from the Georgia Institute of Technology, and an MBA on Strategy and Finance from Cornell University—S.C. Johnson Graduate School of Management. He previously held business and product development positions at Analog Devices and Texas Instruments.

Thinking About DC Power Supplies (Part 2)

Seven Potential Circuits

If you are interested in audio circuits or you need a high-quality, low power "power source" then this article is for you. Following an introduction to multiple circuit types suitable for high-quality audio designs in Part 1, the author now considers seven different circuits for electronic DC power supplies.

By
Frans de Wit

Just a Zener Diode

To begin, we will start with a single Zener diode to regulate the output voltage. To lower the output impedance of the regulator, we supply it with an output capacitor. I will use the same capacitor in all following schematics (**Figure 7**).

It is not much of a high-end power supply, but it is the first step we take. The 650mV$_{PP}$ voltage is already an improvement over the 1V$_{PP}$ noise generated, but not very much. Most of the improvement seen in the Fast Fourier Transform (FFT) will come from the filter characteristic of the series element (R1) and the capacitor (C1), not the Zener diode. Also, the 250mW Zener diode will not survive very long in this environment, although it demonstrates the principle very nicely.

Just a Transistor

It is not possible to build a complete regulator around one transistor. It cannot even be called a regulator as it has no reference, at the least not in the context of this article. This circuit uses a capacitor as its reference. The capacitor is charged by a single resistor that has been adjusted so that

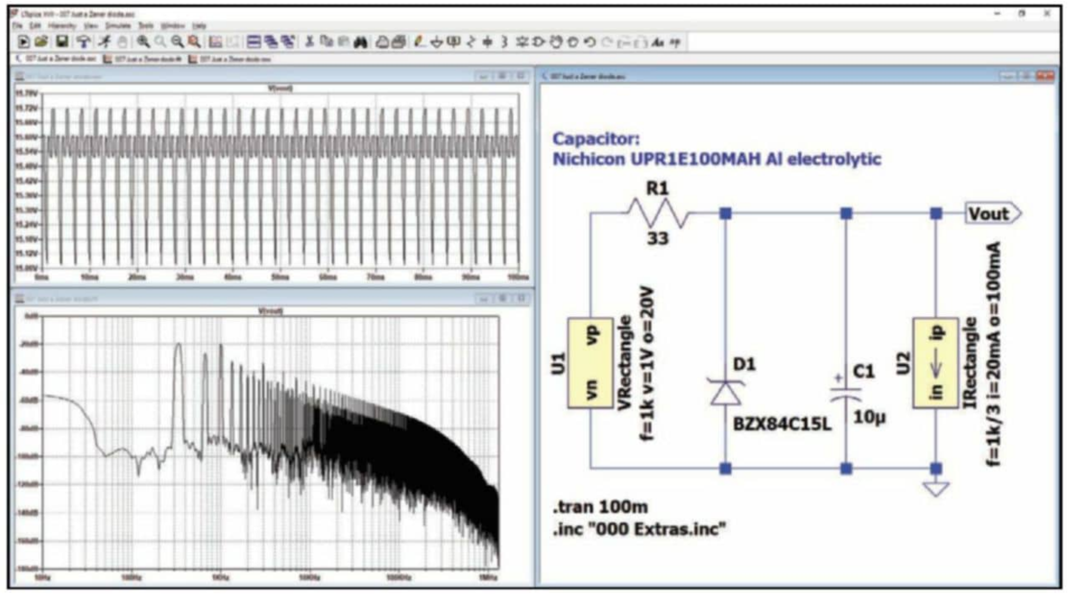


Figure 7: This is a single Zener diode, used to regulate the output voltage.

Tectonic

THE CLEAR CHOICE
FOR NATURAL VOICE REPRODUCTION

Tectonic's Balanced Mode Radiator (BMR), is a unique flat diaphragm loudspeaker that addresses two significant limitations of conventional cone diaphragms; break up and power beaming.

The BMR delivers extended SPL on and off-axis leading to room filling sound with improved coverage for all listeners. Because the BMR is a single full-range solution it eliminates the need for crossovers in the critical vocal region, resulting in excellent voice intelligibility even when off-axis.

BMRs can be found in the market today in:

	BMR	Conventional Full-range	Conventional Multi-way
Wide directivity across full audible range	YES	Maybe*	Maybe*
Smooth sound power response (SWL v Frequency)	YES	No	Yes
Low distortion	YES	Yes	Yes
No crossover needed in vocal region	YES	Yes	No

*Design dependent

TO LEARN MORE VISIT:
TECTONICAUDIOLABS.COM/TECHNOLOGY/BMR-TECHNOLOGY

Contact us at: oem_solutions@tectonicaudiolabs.com or 425.686.7640

Protected by US and worldwide patents

the circuit supplies an average of about 15V to the load (**Figure 8**).

Even though it is not the best, as we will see, it is an improvement. The output voltage shows a peak-to-peak noise of about 56mV, a large improvement over the 650mV that was measured in the Zener diode only circuit. On top of that, the FFT shows an improvement of about 60dB on average and a lot fewer peaks. (Don't even mention the discussion we had. Jan Didden warned me, a "cap multiplier" is not a regulator—and how can I argue with Jan?).

Series, Out of the Emitter

This is a series regulator, the regulating element is in series with the load, and we output from the emitter. For a long time, this was the configuration of choice, all 78*xx* and 79*xx* single chip regulators are based on this arrangement. Nothing wrong with it, you would think. The circuit has gotten a bit more complex. There is now a second transistor that we use for extra gain and feedback. But the circuit is still quite simple and should pose no problems (**Figure 9**).

If we look at the peak-to-peak noise, there is no obvious improvement. After the previous single circuit with 56mV, the 214mV_{PP} noise in this circuit hardly seems to be an advancement. A single -37dB peak at 1kHz remains. Mostly, this shows us that line (source) regulation has improved a lot and load regulation less. But if we look at the FFT, then real progress is evident, the -20dB peaks at 333Hz, 666Hz and 1kHz are gone, and this regulator shows much improvement in the frequency domain. Note that the load varies at 1kHz and the source at 333Hz. All in all, not a bad result.

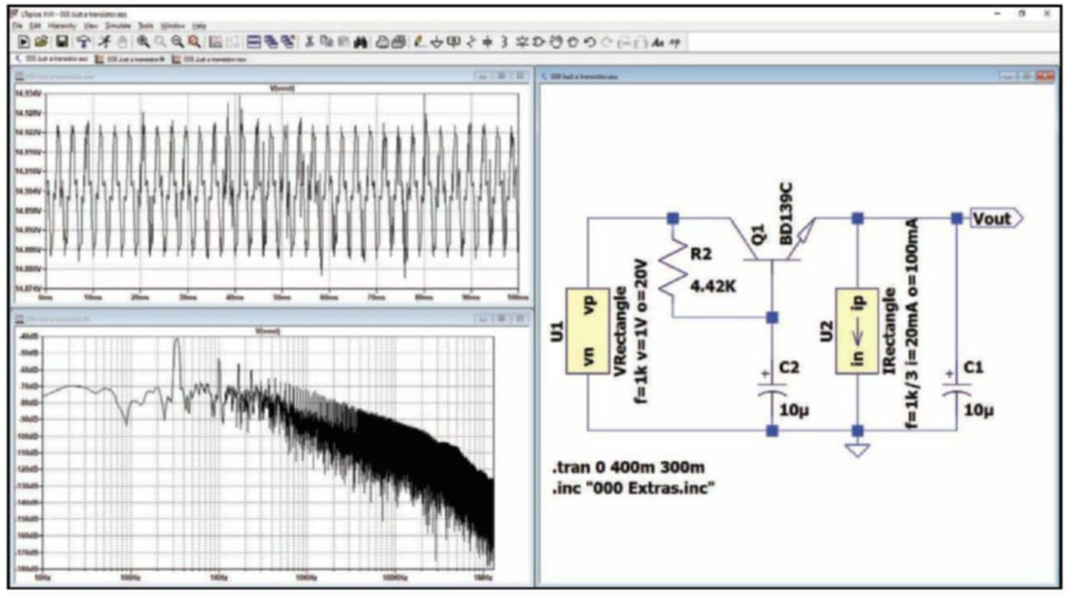


Figure 8: This circuit uses a capacitor as its reference.

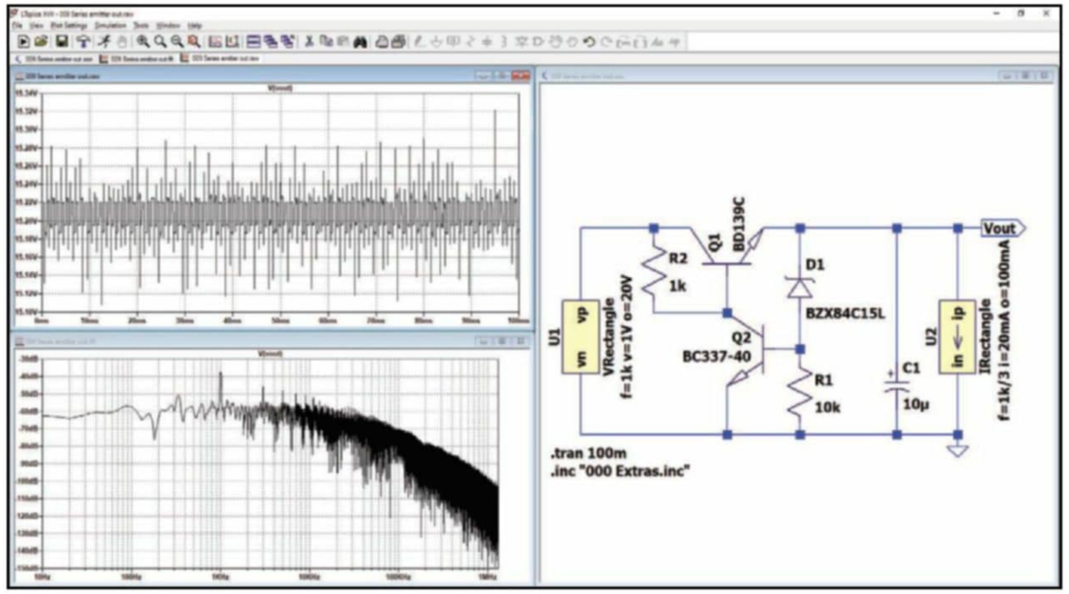


Figure 9: A series regulator has the regulating element in series with the load and we output from the emitter.

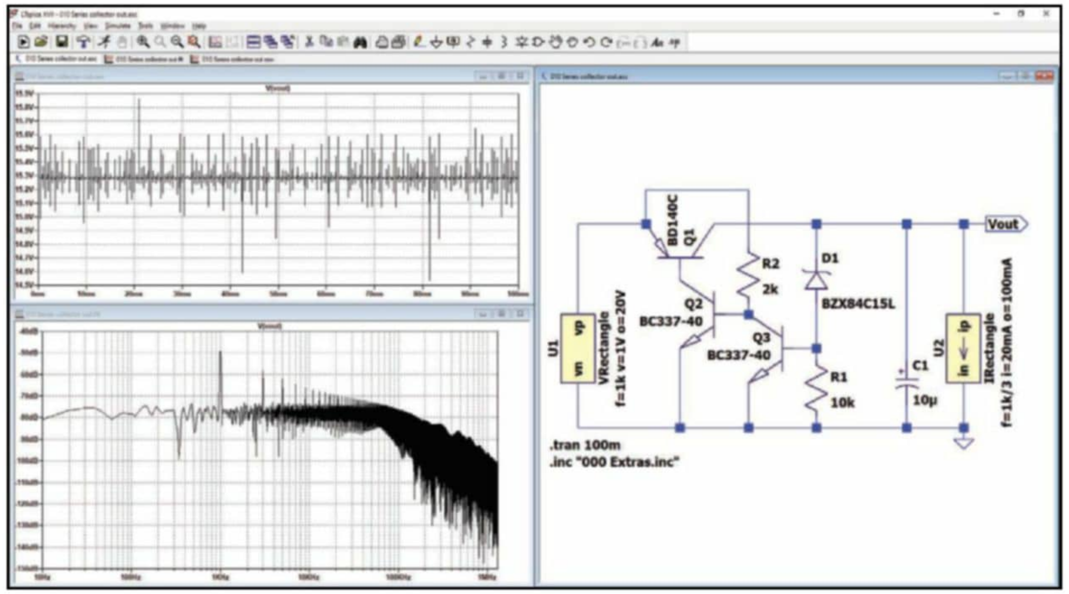


Figure 10: This circuit goes out of the collector. The configuration is also called an LDO circuit, or Low Drop Output regulator.

Series, Out of the Collector

Adding moderate complexity, this circuit goes out of the collector. The configuration is also called an LDO circuit, or Low Drop Output regulator. The feature that makes this type of regulator so popular is that it can operate at a lower input voltage than the previous emitter follower circuit, the emitter follower needing an input voltage that is, at the least, one V_{be} higher than the output voltage. The LDO configuration only needs one V_{ce} (**Figure 10**).

Because the control signal for the output transistor needs to be inverted, the circuit needs one "extra" transistor compared to the previous

emitter follower circuit. The 1.33V_{PP} noise on the output voltage also seems to be no improvement, but as before, we need to look at the FFT graph to see the real data of interest. Where the previous circuit showed an FFT-baseline at around -60dB, this one shows a baseline of about -80dB, not bad at all.

Out of the E or C, that is the question—or maybe not so much. The next graph shows the difference as calculated by the last two simulations. The graphs show some minor differences with a baseline difference of about 10dB. As the LDO configuration is using one more transistor this difference can easily be explained away (**Figure 11**).

Shunt, Grounded Emitter

Now that we have had a look at the series regulators, it time to discuss shunt regulators. These first ones are regulators where the active transistor is in parallel to the load. Let us have a look at a grounded emitter circuit (**Figure 12**).

This is performing better that I expected, as 221mV_{PP} noise voltage on the output signal is very good, especially considering the simplicity of the circuit. The circuit shown in Figure 10 has better line and load regulation, about 15dB, but maybe that can be explained away by having one fewer transistor.

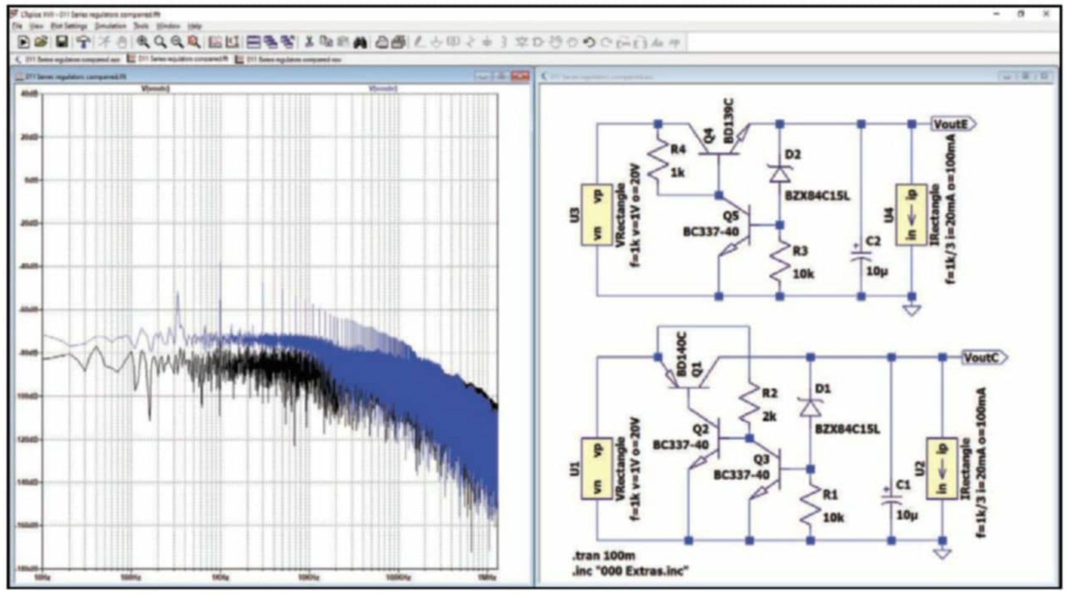


Figure 11: This graph shows the difference between the V_{be} and the V_{ce} configurations.

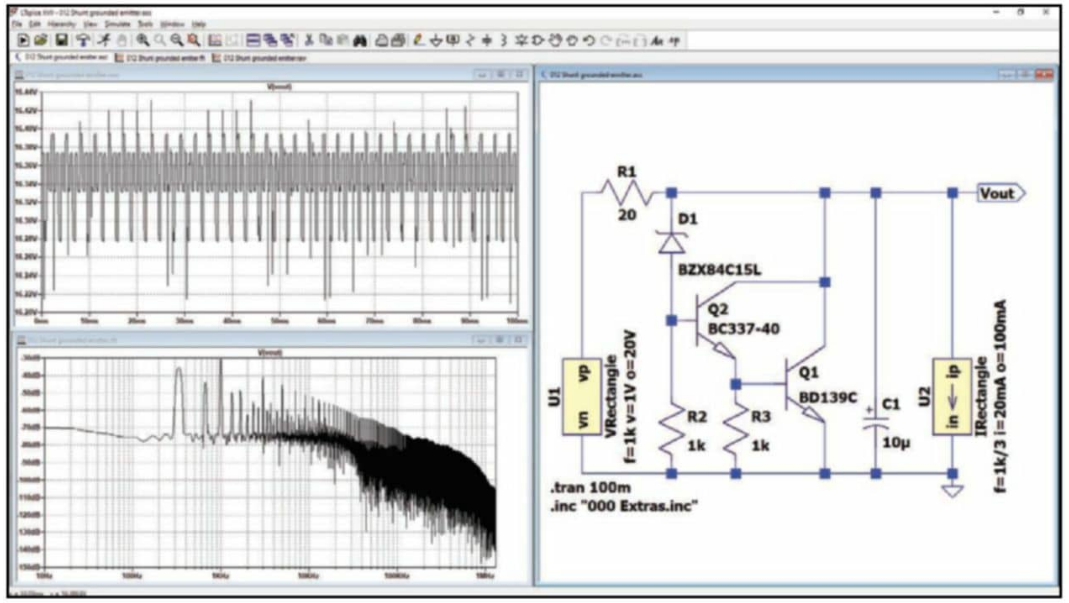


Figure 12: Here is a grounded emitter circuit.

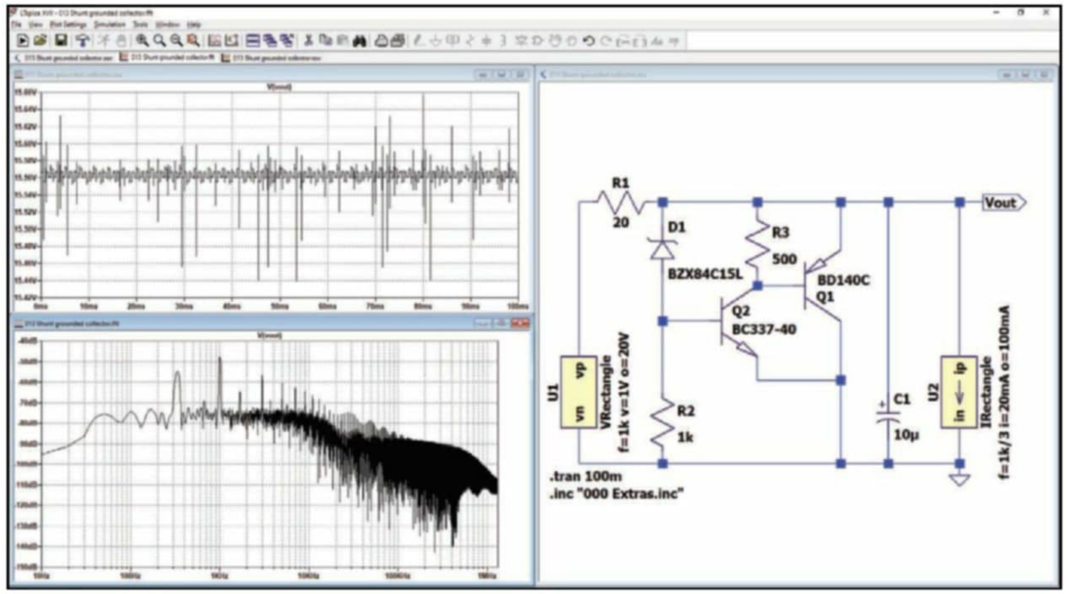


Figure 13: This is the grounded collector shunt regulator, in this example, a complementary feedback pair.

Shunt, Grounded Collector

And then we arrive at the grounded collector shunt regulator. For this I will select a complementary feedback pair, also known as a Sziklai pair (**Figure 13**).

Compared to the previous circuit, we see line and load regulation improved by near 20dB and the 666Hz peak is almost down to -70dB. This might be the best of all the circuits shown here. At the least, we see why shunt regulators have stayed popular through time and technology changes. The one drawback (equally so for the previous circuit) is power consumption. A quick check here will show that to supply 100mA at 15VDC, or 1.5W, this circuit consumes about 3.5W of energy, meaning that more than two times the output power is "burned." Although this is a disadvantage, for both shunt circuits, it is not as bad

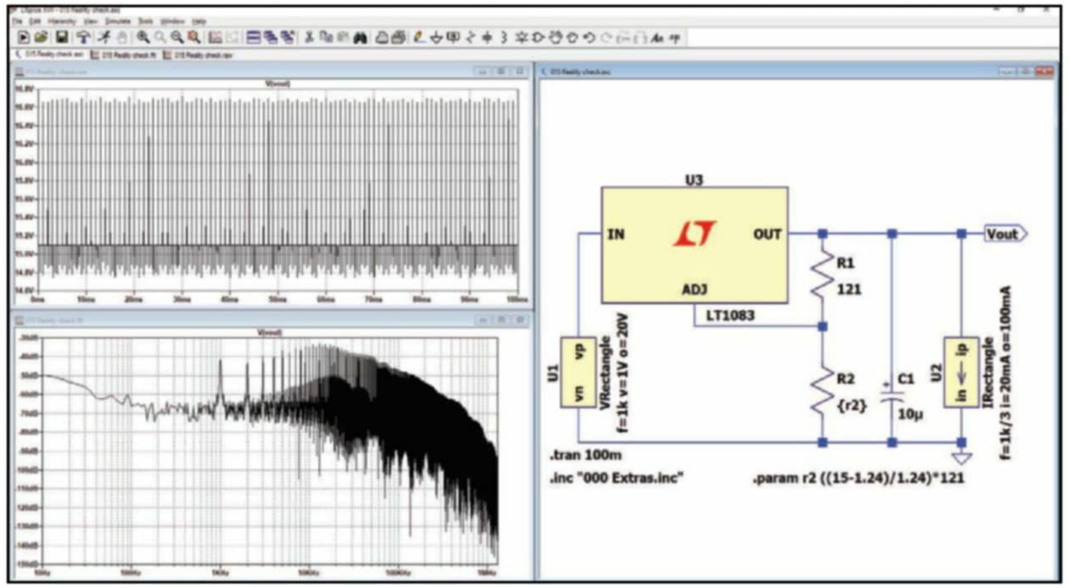


Figure 14: This example shows the LT1083, which has its model supplied with the LTspice simulation program.

as it seems to be. The other circuits are dissipating slightly more than 2W each, and for the type of application for which we would normally use a circuit like this, consuming a near 100mA of current, quality may be a better selection criteria than the 1.5W of power loss.

One other but major advantage of the shunt regulator is that the circulating "noise" or ripple current is confined to the loop between the load and the shunt path (which can be made very small). That means the noise/ripple current is not going all the way back to the primary source. This keeps the current loop area very tight and helps a lot to keep the supply clean.

As noted, there are intrinsic advantages to using a shunt power supply. The most obvious drawback of the shunt regulator remains power consumption. Especially when feeding a low-power circuit (e.g., an audio pre-amplifier), this loss may be acceptable compared to the quality advances that the shunt regulator will bring. A future follow-up article will show such a solution.

Reality Check

Finally, let's have a look at a common 3-pin TO220 regulator. This example shows one that has its model supplied with the LTspice simulation program. The LT1083 is a good solid example of a modern 3-pin regulator (**Figure 14**).

Now this is a true reality check. This commercial product shows peak-to-peak output noise of near 2V, most of it at higher frequencies, and as the FFT shows almost all remaining noise can be found below -40dB. To be fair to the LT1083, it was included here to show that power sources are designed for a purpose. In this case, the LT1083 was misused to make a point. Like the earlier regulators shown in this article, I wanted to demonstrate what merits or problems these circuits exhibit, most likely the source and the load are a bit more demanding than any real-life situation demands.

In Conclusion

There we have it: all the models and the solutions that I wanted to discuss have been shown. Is this the end of the line? Of course not, technology progresses, new solutions create new problems these create new solutions that will, again, create new problems ...

One of the major challenges for the circuits described here is that the demands we are placing on them may be too great for such small circuits—maybe 120dB over 4 or 5 octaves is too much to ask. But, there is no harm in trying and that was what this exercise was all about.

About the Author

Born and raised in the Netherlands, **Frans de Wit** went to a high school to become an electrical engineer. After this he found his first job in electronics retail, selling components over the counter and advising customers on component selection and application. Here he was spotted by a headhunter and invited to work for the Dutch importer for Motorola, Texas Instruments, Hewlett Packard, and many other great brands of the time. There he started his career as a Desk-Sales Engineer. During this time, he helped customers building trains, organs, milk-floats, and particle accelerators (yes, the one at CERN) among other things. Next, he co-founded an IT company, and until 2016 he worked as a programmer and algorithms specialist building operating systems, compilers, and a large DAM system. In 2016, he founded a long-time ambition, an audio research and development company named Signature Origin.

Please download the file(s) that accompany this article series and start experimenting for yourself. Everything that you need can be found in the supplied LTspice and the files. Places such as DIYaudio.com supply a platform for discussing electronics, their applications, and some experiments including the LTspice simulator.

The power supplies shown in this article are skeletons, with an absolute minimum of components—all active components are un-supported and un-compensated. The source voltage is designed to agitate and the load-element is created to provoke. The author has not found a comparable modelled comparison between all these design approaches, and felt that it was time to provide such an overview.

By the way, I plan to continue this discussion with a future article. In it I intend to demonstrate that extraordinary design efforts can create extraordinary results in real-world power supply applications.

Author's Note: Thanks for reading this article. I would like to thank João Martins, Jan Didden, and Ward Maas for their help making this possible, by supporting me to "go for it" and proofread/ghostwrite where needed.

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html

To be able to make fully make use of the files provided with the article you need to install a copy of the LTspice circuit simulator software, which is available from: www.analog.com/en/design-center/design-tools-and-calculators/ltspicesimulator.html

Together with the simulation files comes one include-file, this file will be included with all simulations that will be provided with the article. The file contains a few transistor models and a few Spice directives. Just open the file with a simple text editor, like notepad, to examine its contents,:

.options numdgt = 7 Setup for high-resolution calculations.

.options plotwinsize = 0 This will improve the drafting quality.

Resources

For more information about the Silent Switcher products visit:

www.analog.com/en/products/landing-pages/001/silent-switcher.html

An Experience with Premium Vinyl Records

By
Mike Harkins

Are today's audiophile vinyl records and pressings better? Are today's expensive vinyl reissues "as good as has ever been pressed"? That's what the author wanted to find out when partaking in a simple experiment that discusses legitimate questions music enthusiasts asks themselves.

Photo 1: The author's playback system includes a Hana EH phone cartridge mounted in Clearaudio Concept turntable with Satisfy tonearm, calibrated, and set to 2 grams tracking force; a Puffin phono DSP preamp; a Creative Labs Sound Blaster Zx audio card installed in Alienware Windows-based computer; VinylStudio recording and analysis software; Grado SR125 headphones and Vanatoo Transparent Zero powered speakers.

For several years, audio enthusiasts have embraced vinyl as a preferred source of music. If you have gone to pre-pandemic audio shows or even your local dealer, you have probably witnessed a preference for high-quality vinyl records over other music sources, except perhaps 15 inches per second (ips) master quality tapes. The fact that a mechanical stylus can so accurately pick up and transmit an analog electrical signal that so closely matches that of the original recorded sound is quite astounding by itself. Thomas Edison never envisioned his talking machines as anything more than a tool for dictation and messaging. But here we are in 2022, where audiophiles seem to adore the direct descendants of his cylinder medium—the vinyl record.

I am sure part of the reason for vinyl's comeback is simply nostalgia for a tried-and-true medium, which allows manual selection of records and tracks. But among audio enthusiasts, could it be that vinyl recordings are being made at a much higher level of quality these days? I refer primarily to the limited-edition audiophile pressings. Some claim a maximum of 1,000 discs made per stamper and other means of controlling production quality, such as the use of high-quality virgin vinyl and quality assessments of test pressings. Can such a difference be detected? The answer to that question would involve some experimentation.

In my experience, records are notoriously noisy. Some have rumbles, hum, print-through, or swishing sounds that can come from the master recording or from record wear. But nearly any record will have the "pops" or "clicks" (aka ticks) embedded in the vinyl so that upon playback, these noises mar the musical sounds permanently recorded in the record groove. It is intriguing to note that the same audiophiles who condemn the slightest hint of digital noise, RF noise, dithering, and other faults of digital technology, completely ignore the louder sounds found in vinyl. Admittedly, vinyl sounds are usually instantaneous, impulse-like sounds, and for that reason, they may be less annoying than some other noises. So, the type of noise can make a big difference in our perception of it.

There are several processes involved in the manufacture of vinyl records. Some of them can impart noises, flaws, or defects in the playing surface. For example, the master lacquer must be plated so that it can transfer the analog groove pattern to a negatively grooved tool, which ultimately is transformed through other plating processes to a record stamper, the mold used to make the finished product. Each plating process must be free of dust, debris, and contaminants so that no foreign particles or irregularities (e.g., plating porosity) can mar the groove surface, thus causing a pop or click upon playback. This is difficult to do since contamination is everywhere. Nevertheless, it is surprising that only small defects of this type are audible upon playback of the typical recording.

Vinyl is not indestructible, and we use the hardest known substance in the universe to play it: diamond. Even though phonograph records can supposedly have up to 70dB of dynamic range, most seem to have only 30dB to 40dB and those pops and ticks can make sounds as loud or louder than the program material. It is the nature of the beast.

My focus is on pops, ticks, or clicks, as we refer to them. The reason is simple. It is these artifacts that are the hallmark of vinyl noise. They appear in the manufacture of records, but they are often caused by simple dirt or fingerprints on the records. Such self-induced flaws can and should be cleaned off prior to playing a vinyl record. But it is preferable to handle records so that fingerprints and dust are kept from landing on the record's playing surface. Then, a simple brushing with an anti-static cleaning brush should be sufficient to eliminate the surface dust on the record.

It is possible that crackling sounds or other static-like noises could appear on a vinyl record. But in my experience, most of these are simply an accumulation of pops and clicks, to the point that they are repetitive in a small time period of record playback. Most other noises (e.g., background noises at live recording venues, hum, print-through, and tape hiss found mostly on older recordings) are inherent in the master recording and have nothing to do with the fact that the playback medium is a vinyl disc.

It is also possible that some distortions are inherent in vinyl recordings. To be sure, that may be true, especially when playing the inner groove area of a record. However, my goal is not to determine playback flaws or even recording flaws, but to determine the quality of the vinyl medium itself. Pops and ticks are the usual flaws found in vinyl. They exhibit as an impulse or a quick cycle of perhaps a millisecond.

The intent of these tests is to show that audiophile vinyl recordings these days are good quality. I am not expecting perfection, but I am expecting new vinyl to outperform the traditional $6 LP bought in 1977. Having been a quality professional at several large tech companies for more than 30 years and, after gathering the right information, I hope to demonstrate quantitatively and qualitatively, the quality of these recordings.

My mission was simple: Procure a few of these superdiscs and measure their performance in terms of vinyl noises, such as pops and clicks. Are any of them of such poor quality that they are unacceptable? Are they worth their $35 to $150 (or more) price tag? For that price, they should at least be the equal of the CD version or the hi-fi quality streaming audio version. If not, why would the listener not opt for a superior format except for nostalgia's sake?

I selected several recordings to examine. I picked one recording that I already had in my collection and others that I did not. The recordings I chose were *Simple Dreams* by Linda Ronstadt, one of her most popular records from 1977. I already had this recording, although not an audiophile version, in my record collection. I also selected *Come Away with Me* by Norah Jones, her debut album and a great one from 2002. I added to these Lyn Stanley's *London with a Twist/Live at Bernie's*, and Nina Simone's *Pastel Blue*. I did not intentionally choose female vocals for these tests. I merely wanted to choose something that I could easily evaluate. All but the last of these are on audiophile labels. These four albums on six discs were used to measure various aspects of vinyl noise. In addition, I examined two more single-disc audiophile recordings to gauge, without actual measurement, their overall sound and noise quality. These albums were Dave Brubeck's *Time Further Out* and Nina Simone's *I Put a Spell on You*.

Photo 2: The author's first listening test was conducted with the 45-rpm recording of Linda Ronstadt's *Simple Dreams* (Cat. Number 753088107474) 200-gram 2LP 45 RPM Analogue Productions reissue. Mastered by Doug Sax and Robert Hadley, plated and pressed by Quality Record Pressings.

My Home Setup

I archive vinyl records into a digital FLAC file format at 24-bits and 96kHz sampling. This FLAC format allows extraction of the best sound quality from my records. I have no idea why I am doing this since I am a Tidal subscriber and most anything I want can be found there. But nevertheless, I like to extract information from vinyl just to see if I can get a decent recording from the records and place them on my Roon Nucleus+ core server. The results are usually quite good. From there, using Roon, I can stream either locally recorded vinyl or

Photo 3: Examination of the spindle hole on the Linda Ronstadt album showed it to have a clearance of about a millimeter.

a selection of digital treasures from CD quality to higher resolutions or any of the DSF (SACD) formats. I can thus compare captured vinyl recordings to those I stream on Tidal or to high-quality audio files on my hard drive.

Photo 1 shows my playback equipment, which includes:

- Hana EH phone cartridge mounted in Clearaudio Concept turntable with Satisfy tonearm, calibrated, and set to 2 grams tracking force
- Puffin phono DSP preamp, set to RIAA flat response
- Creative Labs Sound Blaster Zx audio card installed in Alienware Windows-based computer
- Recording and analysis software: VinylStudio
- Grado SR125 headphones and Vanatoo Transparent Zero powered speakers
- Playback was also done on Legacy Signature SE speakers bi-amped with Outlaw Audio power amps

So, I do enjoy listening to vinyl, but in digital source format. One real benefit of this is not having to go through certain rituals, such as record or stylus cleaning, and so forth whenever I want to play a record. I can detect no audible degradation of the sound in digital format. Some purists may eschew digital in all its forms. But I might mention that our enjoyment of superb 4K and 8K video formats are thanks to digital encoding of television signals. Yes, digital has done wonders for that format. Why can't it do the same for a relatively much simpler medium: Audio? But I digress. My attention here must turn to vinyl.

First Experiment

Although the biggest problem with vinyl phonograph records is noise, I will make note of any other defects or flaws found in these recordings. My criteria for a flaw would be any audible noise that is not part of the music. A defect would be something considered unacceptable enough to cause me to request a replacement disc. That disc would be described as defective, having one or more unacceptable defects.

Do these new recordings have little to no noise? The answer to that question would come from my first listening to the 45-rpm recording of Linda Ronstadt's *Simple Dreams*. The full LP is contained on two 200-gram 45-rpm discs (**Photo 2**). Even at 45-rpm, these discs only had about the first 2" of vinyl from the disc's periphery used for the actual recording. The lead-out groove area was as big as the recorded space on the disc.

A cursory examination showed the discs to be clean and well-made with no apparent anomalies other than a slight warp on the first disc. Even small imperfections can be picked up by the playback stylus, but many are not audible, including most record warps. Upon playing side one, I noticed a few things. The software I used showed this record to be compressed to the point that I could record it with peaks consistently topping out at -0.4 dB. That was about the same as a previously recorded LP version of this disc, which dated back to its release date. Compression should make overall signal-to-noise better.

Second, the stylus wavered back and forth about 1mm or so as it played. There was an eccentricity. Examination of the spindle hole showed it to have a clearance of about a millimeter (**Photo 3** and a close-up view shown in **Photo 4**). I could center the record on the spindle manually, but that did not eliminate the eccentricity entirely. Audible effects of this were not apparent, but if pure tones were on the record instead of music, we might hear some evidence of wow.

The sound quality was quite good on side one.

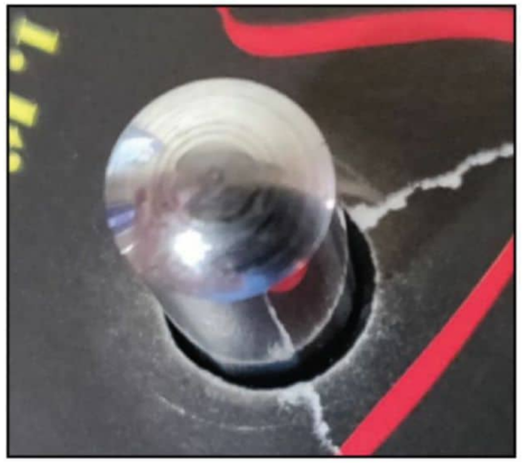


Photo 4: Here is a close-up view of the clearance between the hole and the spindle.

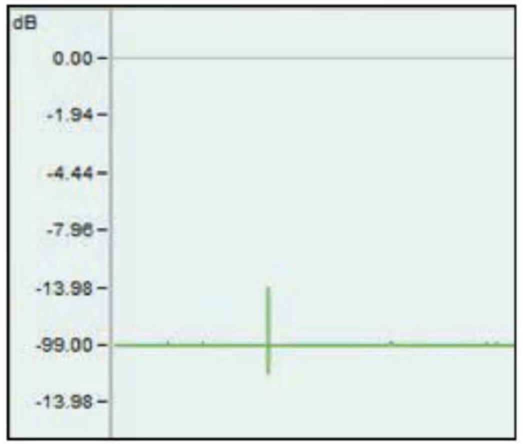


Figure 1: A tick between album tracks on Linda Ronstadt's *Simple Dreams* was clearly audible to the listener.

There appeared to be no over-emphasis of any part of the signal spectrum and it sounded much better than my 1977 LP version of this recording, which I admit did have some record wear. Signal-to-noise was also quite good. But the best measured signal-to-noise as evidenced by my VinylStudio software was 42dB. This is a far cry from the potential of 70dB and certainly nowhere near a CD's 90dB potential.

And of course, there were a few of the inevitable "ticks" that are an unending artifact of vinyl playback. There is one (**Figure 1**) between the first and second tracks of Ronstadt's recording. Notice that this tick has a magnitude of about -14dB. Most assuredly audible! The duration of such imperfections causes them to be not very annoying and if drowned out by program material, they would be mostly inaudible.

At this point, I'm sure many in the "vinyl forever" camp are saying, "Didn't you clean the record first before playing it?" Good question. But I figure if I just paid $35 for two audiophile-grade 45-rpm vinyl discs, they ought to be pristine from the factory and not require anything more than a superficial brushing. Boy was I wrong!

Side 2 of this LP contained sufficient noise content that I would call it defective. A large splotch of unidentified material was visible to the naked eye when closely examined. When I played Side 2, I noticed noise occurring in the first 30 seconds of playback. After stopping playback, I got out my magnifier and looked carefully at the track I was playing. Sure enough, there was a large imperfection that escaped the scrupulous eyes of the factory inspectors. Of course, I know from my many years in the quality profession, that inspection is at best, a way to remove only a portion of the defects made during production. Based on this one imperfection alone, the defect rate of my sample of nine discs (including a replacement) is 11%.

Figure 2 shows the noise appearance on my VinylStudio program. **Photo 5** shows what the defect looked like under magnification. Unlike cosmetic defects, any such defect in a vinyl record would cause a noticeably loud noise upon playback involving more than one turn of the record on the turntable. No, this was not a pop or tick, it was a loud scratching noise, quite audible to anyone with a working set of ears. Figure 2 also shows the original audio track, in green. VinylStudio, using a filtering algorithm, was able to take out most of the noise from the signal allowing the green, non-noisy signal to emerge. Notice that the noise is much larger than the original audio track. The right channel (not shown) clipped at 0dB due to the noise.

I used a good dose of VinylStyl liquid record cleaner and a velvet brush to try to remove this, but it did nothing to eliminate or even reduce the magnitude of this defect. Notice from the scale at top of Figure 2, that the noise covers about 7ms of time. The noise repeated through several revolutions of the record, but at much diminished volume. I asked for, and promptly received a replacement disc, which was of excellent quality. I included that disc in the nine samples used for these tests.

Despite these noisy events, sound quality of this album is quite good. The vocals, the guitar, the drums, bass, and the music in general sound right

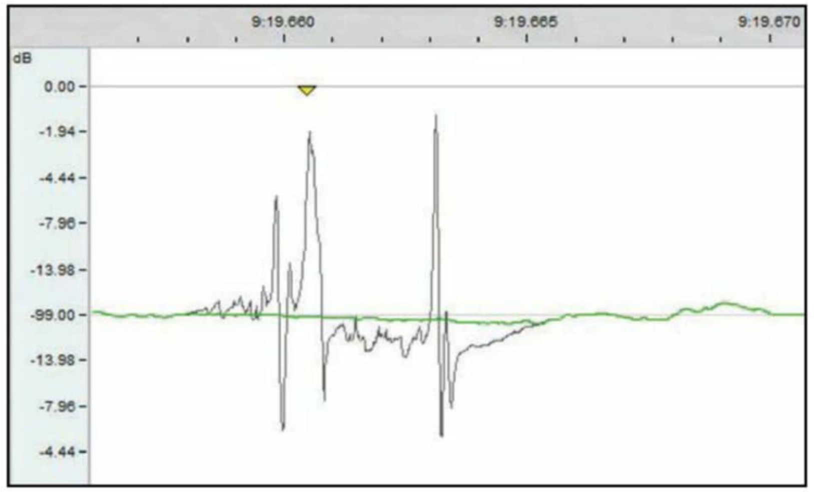


Figure 2: This graph shows a noise anomaly on side 2 of *Simple Dreams*.

to me. The only downside seems to be the noise and it is rarely audible except in a few places, such as the one shown in **Figure 3**.

Second Experiment

Now on to Norah Jones, a vocalist with an angelic voice. Her album was recorded at 33-1/3 speed, and is a more conventional LP (**Photo 6**), although 200 grams of vinyl is used to make the record along with similar quality control as the Ronstadt record since they are on the same label. Did they do a better job here? To start, the center hole was a much tighter fit (**Photo 7**). Although unmeasured, the play was certainly less than 0.1mm since I could barely detect any. There was no noticeable eccentricity of this record.

Sound quality was superb and the dynamic range noticeably better than that on Ronstadt's recording. Jones' voice was clear and up-front with no noticeable tweaks or heavy reverb or echo effects. Guitar, bass, and drums sounded real. In my opinion, this is a better recording than the Ronstadt recording, probably due to a better master, no Aphex processing, and better vinyl pressing.

The vinyl was not perfect though. I could hear an occasional noise or two, but they were rare and unobtrusive. Notable among all the tracks on this album was "I've Got to See You Again" on Side 2. This was the second track on that side, and I really enjoyed the nice mix of instruments, including, a gentle fiddle, some light tambourine, and highly supportive drumming. Most noticeable, was the almost total lack of noise. In some gap areas, I noticed over 48dB of silence as shown by VinylStudio's decibel meter. Of course, compared to CD quality, this is nothing to write home about, but it is particularly good for vinyl.

To understand the amount of noise between tracks, I used VinylStudio to zoom in to 50× magnification of the signal (**Figure 4**). When I did that, I found that most of the noise was very low frequency, around 10Hz and therefore inaudible. This frequency was likely caused by record warp. The rest of the noise seen superimposed on the low-frequency wave in Figure 4, I judged to be approximately 60dB to 70dB below maximum signal level, and therefore inaudible.

So the bottom line, this album is noticeably quiet. It was certainly quieter than the 45-rpm Linda Ronstadt album, and that really put a smile on my face. I am sure on an A-weighted scale, which considers our hearing sensitivity, the noise would be around 70dB lower than the maximum signal level.

One More Experience

A colleague suggested that I try a "direct-to-disc" recording, which he claimed was excellent in both sound quality and performance. While I won't comment on the performance, I did evaluate the sound quality. The album in question was Lyn Stanley's recording, *London with a Twist/Live at Bernie's* (**Photo 8**).

Photo 5: Here is a magnified view of the defect on Side 2 of the Linda Ronstadt album.

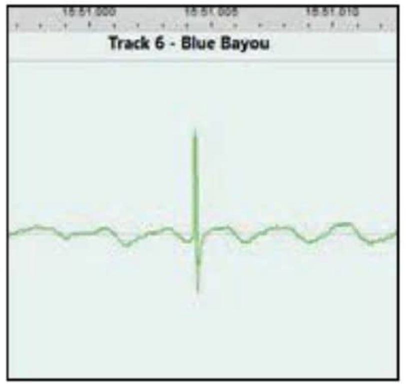


Figure 3: A tick in the song "Blue Bayou" with magnitude much greater than the program signal.

Photo 6: Norah Jones' *Come Away With Me* album was recorded at 33-1/3 speed, and is a more conventional LP (Cat. Number 753088004216) 180-gram vinyl pressing by Quality Record Pressings. Mastered by Kevin Gray at Coherent Audio from the original source recordings.

Photo 7: The center hole of Norah Jones' *Come Away With Me* album was a much tighter fit.

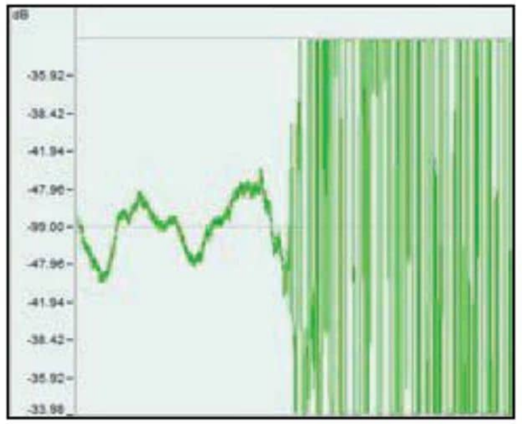


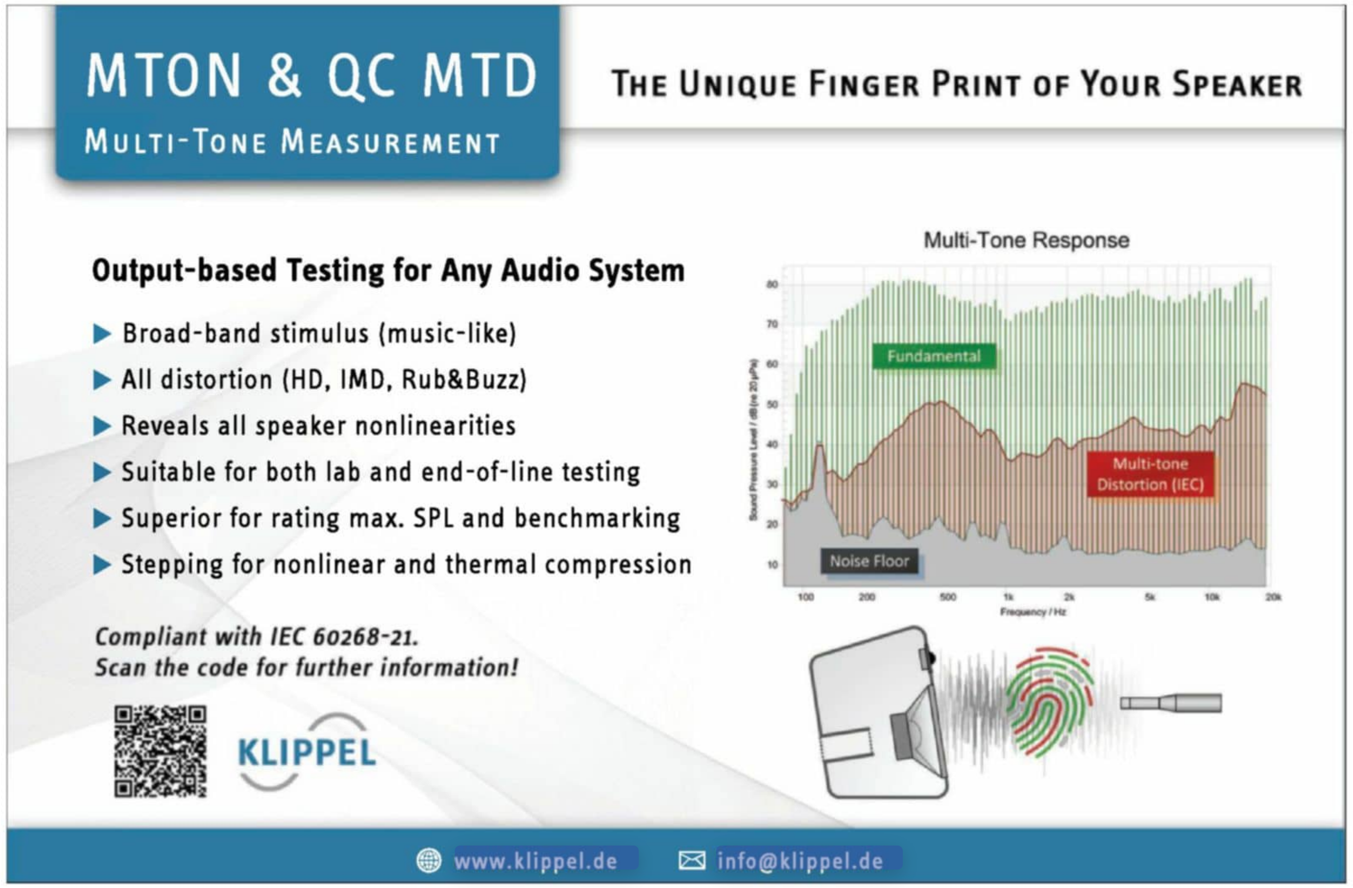
Figure 4: Magnified 50× intertrack noise is shown on the left side of this signal display.

This is a double album recorded at 45-rpm. It is on red translucent vinyl. My listening tests revealed a very pure sound with practically no noise. The VinylStudio software I used for recording and digitizing the disc did pick up some ticks, however. VinylStudio is overly sensitive and reported over 5,000 potential clicks but attempted to repair only half that number due to percussion or brass protection. When I looked at a blow-up of the audio signal, it appeared that nearly all the clicks reported were down at least 40dB or so and were thus, totally inaudible due to their brief duration. In fact, the loudest such noise I detected was found on Track 5, at 16:13 into the recording (**Figure 5**).

As you can see, its magnitude is about 8dB down, but surprisingly, it was nearly inaudible. It could be heard as a very sharp, but brief tick in the left channel. Although Figure 5 doesn't show it, most of the audio at this point in the album was quite loud, rendering the noise shown nearly inaudible.

Notice also, that the entire spike is only about 0.5 ms long. When listening to the album, I also heard a spike of the same magnitude on Track 6, "You Never Can Tell," at about 36 seconds into the track. This one was not picked up by my anti-click software, but a scan of the audio proved it was there, sure enough. It was not a glaring defect, but a flaw, nonetheless.

I am not a believer in scrubbing new albums with cleaning solution to remove debris. For the premium prices these recordings command, I see no need for it. But to see if cleaning did anything to the minor flaws in this record, I grabbed my VinylStyl velvet cleaning brush along with a good dose of VinylStyl liquid cleaner to see if it would do any good.



I know there are some who would not use anything less than one of those ultrasonic or super-mechanical machines that cost hundreds of dollars to clean records. But I think my solution should do something if indeed, the imperfection is removable. Alas, after two cleanings with VinylStyl, the imperfection was still there with no change in magnitude.

This is not to say that the album is not superb. It is near the top of the vinyl heap. But the imperfections of vinyl, unless they are just superficial dirt, cannot be cleaned off with record cleaning solutions and brushes.

The bottom line for this album is that its vinyl quality was surprisingly good. Coupling that with the superb direct-to-disc recording makes it an even more satisfying experience. I have only two other direct-to-disc albums, which do not have such a quiet surface. One is Dave Brubeck's direct-to-disc album, *A Cut Above*, featuring Brubeck and his three sons in the new Brubeck quartet. The other direct-to-disc album is a Telarc recording *Direct from Cleveland* performed by Lorin Maazel and the Cleveland Orchestra. These albums date from 1977 and 1978 and neither were ever as clean sounding as the Lyn Stanley recording from 2019.

Lyn Stanley's album was quite expensive however, proving that you can get good quality, but perhaps you must pay big bucks to get it. And for that price, you also get a personalized signed copy from Stanley herself plus a bonus CD. But I think it is important to see if a normally priced record can also sound good without vinyl imperfections.

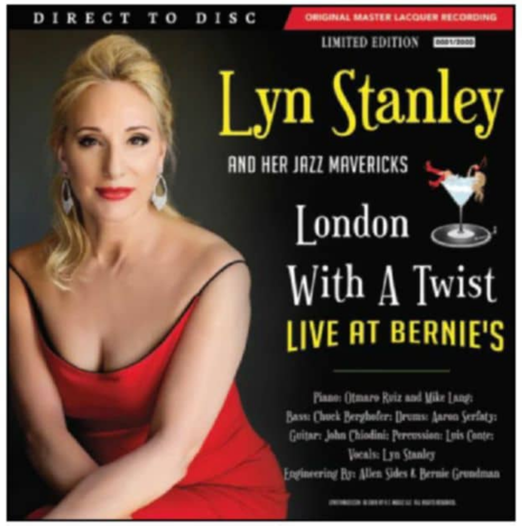


Photo 8: The author also tried a "direct-to-disc" recording, Lyn Stanley's recording, *London with a Twist/Live at Bernie's*. This Limited Edition album, a direct-to-disc recording at Bernie Grundman's Mastering Studio in January 2019, is also available in SACD. It includes twelve CD tracks and 24 DSD tracks. https://lynstanley.hearnow.com/lyn-stanley-london-with-a-twist-live-at-bernie-s-sacd

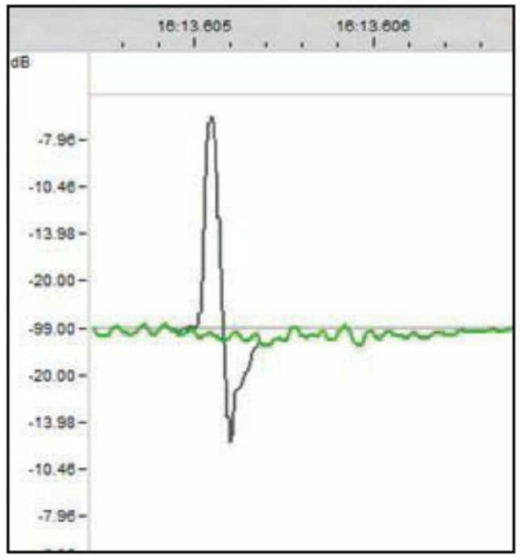


Figure 5: I found this noise spike on Lyn Stanley's D2D recording, *London with a Twist/Live at Bernie's*.

Another Comparison

A friend recommended that I obtain a vinyl copy of Nina Simone's classic album, *Pastel Blue* (**Photo 9**). The album is available on Amazon for around $18 including shipping. So, I took the plunge and bought a copy. This album is on Philips and was pressed in the Netherlands on 180-gram vinyl. Apparently 180-gram or 200-gram vinyl are the watchwords for good quality these days.

To say the least, I was not disappointed. The only annoyance with Nina Simone's album was an audible hum on the first track, "Be My Husband." I checked Tidal and the MQA version has a 60Hz hum, but the CD-quality version does not.

Other tracks lacked this hum for the most part, but that one was annoying. Fortunately, VinylStudio has a hum-removal feature that can completely remove such artifacts, so I later took the time to do so.

This recording is quite good in capturing Simone's sultry voice. The vocals sounded a bit compressed to me, especially compared to the high-quality rendering heard on the Lyn Stanley disc. I also tend to dislike some artificial reverbs, such as the one in this recording. But the recording does a good job on the band's instruments and timbre.

Since the recording dates to 1965, I am sure some degradation has taken place and the original recording may not be state-of-the-art for that era. But it is still quite amazing what they were able to do back then. This record, except for the hum, shows it off quite well. The final track, "Sinner Man," is full of hand claps and a stimulating gospel rhythm that is sure to get you off the couch.

As to quality of the vinyl, it was near-perfect. And this is not a true audiophile recording since It is priced in the mainstream area for records these days. It is also not on an audiophile label. But despite the good quality of this album, it was not perfect.

There were a few minor flaws. For example, **Figure 6** shows an impulse (click) that I found. It is less than -34dB in magnitude and less than 1ms. Thus, it is barely audible. The rest of this signal is intertrack noise between Track 3 and Track 4 on the album. This album is, indeed, noticeably quiet. It is better in that regard than the Lyn Stanley direct-to-disc, despite the Stanley album costing nearly 10 times the price of this one.

I then scanned the entire album. I was only able to find one tick that had a more audible magnitude and that was on Track 8, as illustrated in **Figure 7**. Compare it to the higher noise spike on the Lyn Stanley disc (Figure 5). It was clearly audible as a very brief tick.

Such tiny imperfections are inherent in the vinyl medium, and I will admit, they do not seriously detract from the music. Their presence, when heard, simply tells you that you are listening to a vinyl recording. Without equivocation, such imperfections are nowhere near as annoying as the tape hiss that plagued the early days of tape playback or the hum present on the first Nina Simone track

Final Thoughts

That was the extent of my personal testing, measurement, and evaluation of several vinyl records of mostly audiophile-grade. Beyond these personal tests of vinyl record quality, I checked for comments from others in the evaluation section record reviews on the Internet. One caught my attention. It was the album *Smetana's Má Vlast* by the Bamberg Symphony. This is a direct-to-disc recording with production limited to 1,111 copies worldwide. One reviewer rated this album an 11 for sound and an 11 for music on a scale of 1 to 10.

But when I saw a customer of that album score it a 1, I wondered what was wrong. The customer's review in a nutshell: "Great music; lousy pressing." This is a direct-to-disc recording and should be extremely high-quality, but apparently, at least one customer did not get his money's worth.

If only 1,111 were pressed and there was one defective, the defective rate would be less than 0.1%. But I suspect that if one person found a defective disc, perhaps some others did too, just as I did with another audiophile recording.

What conclusions can we draw from the observations made in this experiment? I think there are several:

- Modern audiophile pressings are mostly very good to excellent quality based on lack of surface imperfections (audible pops, clicks, crackle sounds)
- All vinyl recordings have at least minor pops and clicks, most of which are inaudible due to their small magnitude, brief duration, and possible interference by program material.

- Record cleaning removes surface dirt but not imperfections molded in the vinyl (most pops and clicks).
- Based on a sample of nine discs, I found one defective, yielding a defect rate of 11%. This is based on a small sample, but it implies that defects could be around 10% or so, even for audiophile recordings.
- Quality of audiophile vinyl recordings may not be better than good pressings that are not on audiophile labels or sold at premium prices.

Photo 9: Nina Simone's classic album, *Pastel Blue*, is on Philips and was pressed in the Netherlands on 180-gram vinyl.

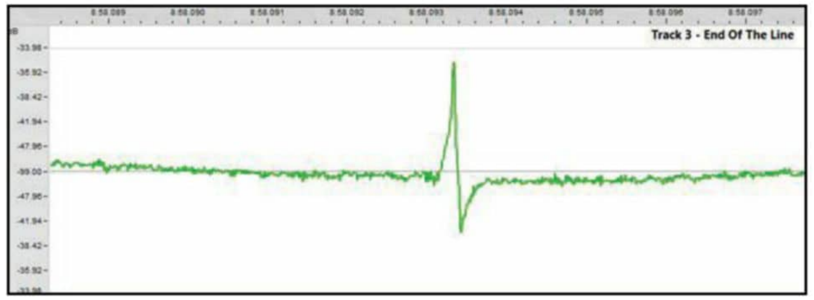


Figure 6: I found intertrack noise on the Philips recording of *Pastel Blue* by Nina Simone.

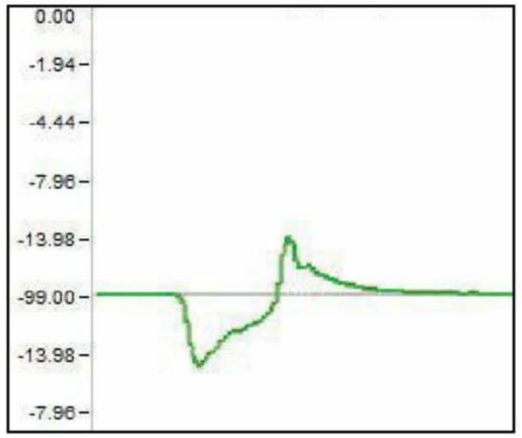


Figure 7: A 1ms blip on Nina Simone's *Pastel Blue* album, Track 8 was clearly audible at -14dB.

This is based on the one non-audiophile recording used in this test.

- Some older recordings can have hum or other annoying sounds, such as print-through from the master tape.

Undetermined from this experiment are the following:

- I could not determine, for certain, if new vinyl is better than vintage audiophile releases of 40 years ago because I had none in completely new condition. However, I know from experience that those earlier recordings were less than perfect and always had one or more audible flaws.
- Although 45-rpm recordings should offer better playback performance, I could determine nothing about their superiority, and they are not immune to the pops and clicks found on all vinyl.
- The quality difference between audiophile recordings and standard pressings is not known even though some non-audiophile pressings are the equal of the premium pressings.

So, the question of whether high-quality vinyl recordings of today are better than ever remains open. The sound is certainly better than the $6 pressing of a record purchased in 1977. Of course, the sound quality of record playback depends on many things. And I only evaluated the noise aspect of record playback, not overall sound quality.

So, what can one conclude from looking at a few sample albums? I guess you must make the fundamental assumption that most audiophile recordings costing $35 and more would be as good quality as these, but perhaps not any better.

If we make that assumption, I will say that audiophile-grade vinyl records are still nowhere near perfect, but they are quite good. Sound quality is on par with the best digital recordings. However, noise in the form of impulse noise, is certainly not as good as digital recordings that completely lack such anomalies.

In addition, putting on my quality hat, I evaluated a total of nine discs, counting a replacement, and one had a definite defect. That is a defect rate of 1 out of 9, or 11%. But since the sample size is so small, what can we say about the likely quality level of a large quantity of these discs based on these observations?

My first assumption in making such an assessment is that these discs were drawn randomly. If we also assume that 10% of discs are defective (which is not good), what is the probability that I would find one defective in a sample of nine discs, as I did?

The answer to that can be found using the binomial probability model. It concludes that there is an 85% chance that I would find one or more defective in the nine samples I choose randomly. That means that if the product is 10% defective, 85% of the time, I would find one or more defective discs in such a sample of nine. So, it is possible that the overall defective rate is at least 10% based on my assumed random sample.

Of course, these are probabilities, and it is possible the overall defective rate is only 1% or less. But if so, the probability of my finding one defective in nine would be about 9%. This means that if I randomly select nine discs from a large quantity that is 1% defective, nearly once in every 10 times I would find at least one defective. This is a low chance, but it shows the possibility that true defect rates for modern vinyl recordings could easily be in the 1% range or even less.

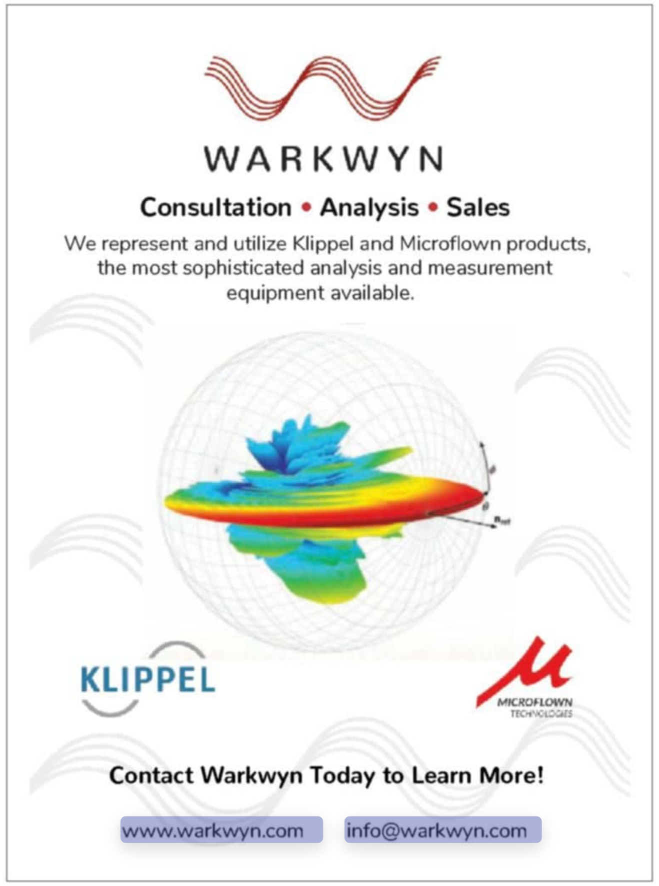
Therefore, good quality vinyl is likely to be available with a defect rate possibly in the 1% to 10% range. An Internet search confirms that other vinyl lovers have seen defectives in their purchased recordings.

So, it is reasonable to conclude that the state-of-the-art in vinyl record production still produces defects. That said, I also found out that manufacturers of these recordings stand by their products and are happy to replace any that are truly defective. That is good news if the defect rate is in the range mentioned.

For those who love vinyl, please keep playing it! Remember, it's love of the music and good sound that really matters.

About the Author

After earning a degree in electrical engineering in his hometown, St. Louis, MO, **Mike Harkins** worked for an aerospace company for five years before switching to manufacturing engineering and quality assurance in the automotive and tech industries. He devoted his career to making improvements in the quality, reliability, and manufacturability of products. His work extended into the service sector more recently, where he helped manage and improve call center quality operations for a large tech firm. Mike retired from corporate work several years ago and has since devoted much of his time to listening to, reading about, and writing about audio. After attending several pre-pandemic audio shows, a 2019 letter to a well-respected audio publisher led to Mike's first articles about two of these events. He has also done blog work for a local audio store. Mike likes to write almost as much as he enjoys learning about and listening to high-end audio. He enjoys attending many musical events throughout the year including live jazz, chamber music, symphonies, classical guitar, and other live performances, especially since most have returned as in-person events with the waning pandemic. He is a senior member of the American Society for Quality (ASQ) and is a member of the Austin Astronomical Society. He lives in Austin, TX with his wife, Theresa.



OMNI Tower VII

An Ominidirectional Tribute

Our veteran speaker builder ventures into omnidirectional territory with a twist. He proposes a floorstanding speaker design that relies on its four sides and is based on a concept devised by the late David B. Weems, using eight drivers, including two Piezo tweeters.

Photo 1: Front and rear views of finished Omni Towers

By
Ken Bird

Omnidirectional speakers are like raw fish delicacies, you either love them or loathe them. I like them, the speakers that is, but I prefer my fish well cooked. Omni speakers received a bad rap from audiophile purists when Bose introduced its 901 System in 1968. The system had nine 4" drivers controlled by a proprietary equalizer and relied on wall reflections to spread the sound throughout the room. It was an attempt by its designer, Amar Bose, to emulate the sound of the concert hall, most of which is reflective depending on the listener's seat position.

Critics claimed the 901 lacked detail and that imaging on vocals was poor. Some person even came up with the slogan, "Bose, no highs no lows." Critics aside, the product helped make Bose the premier audio company it is today. This writer heard one shortly after it was introduced and was impressed with its natural realism, especially on massed voices, choirs, and organ pieces. The 901 was one of the longest lasting consumer audio products having been discontinued only in 2016, and they are still being sought by collectors and audiophiles today.

This project is not to be construed as a Bose knockoff. Bose tends to be touchy about the 901 and past speaker builders have received some letters from its legal department when they tried emulating Bose products. It does use some reflective aspects, but it relies mainly on the four sides of the enclosure to spread the mids and highs around and equalization, or not, it is the choice of the user. Also it incorporates separate tweeters and woofers for the extreme ends of the audio spectrum and contains only a kernel of the Bose concept. The design, like many of our DIY projects today, is based on a concept developed by the dean of speaker building, David B. Weems. It has been updated to today's understanding of speaker systems and driver technology. Each tower holds 4" midrange speakers, two 6.5" woofers, and two Piezo tweeters (**Photo 1**).

Photo 2: The system drivers area is a 4" poly type with a 6.5" woofer and Electrostatic Piezo tweeters.

Photo 3: A mock-up test box stuffed with polyester was used to determine the matching of the 4" drivers with the Piezo tweeters.

The Drivers

The speakers chosen for this system are from the Dayton line sold by Parts Express and include the Model PC 105-4 4Ω, 4" "full-range" drivers. The woofers are the Model PC 165-8 8Ω, 6.5" drivers. The tweeters are the GRS Model 1005 Piezo 3.5" horn (**Photo 2**). See more about using Piezo tweeters in the crossover description. The speakers are wired to produce 4Ω impedance that most solid-state or tube amps can handle easily.

Mock Up and Tower Construction

I made a mock up of the upper chamber from layers of heavy cardboard (**Photo 3**), which I used to test the general function of the four midrange drivers with the Piezo tweeters and the crossover circuit. I tested the various changes in crossover components using the RTA system and listening by myself and with other "critics.".

I constructed the towers using 48" lengths of 1 × 8 select-grade pine boards, which are free of "knots" and are denser than common pine lumber. MFD can also be used as well as any hardwood that DIYers desire. The tower is divided into two sections with the lower half being the woofer enclosure and the upper half housing the tweeters and midrange speakers. The crossover network is divided, with one section mounted in the lower chamber behind the woofers and the other in the midrange tweeter chamber. The bases were made from 1/2" plywood and were sized to prevent tipping of the enclosure (**Figure 1**).

I started construction by marking the center lines of the boards and then placing the drivers on the boards in their respective positions. Use the mounting holes to determine the center of the driver by drawing an X between the holes. This will give you the pivot point for the circle jig (**Photo 4**). I cut the necessary number of holes using a router

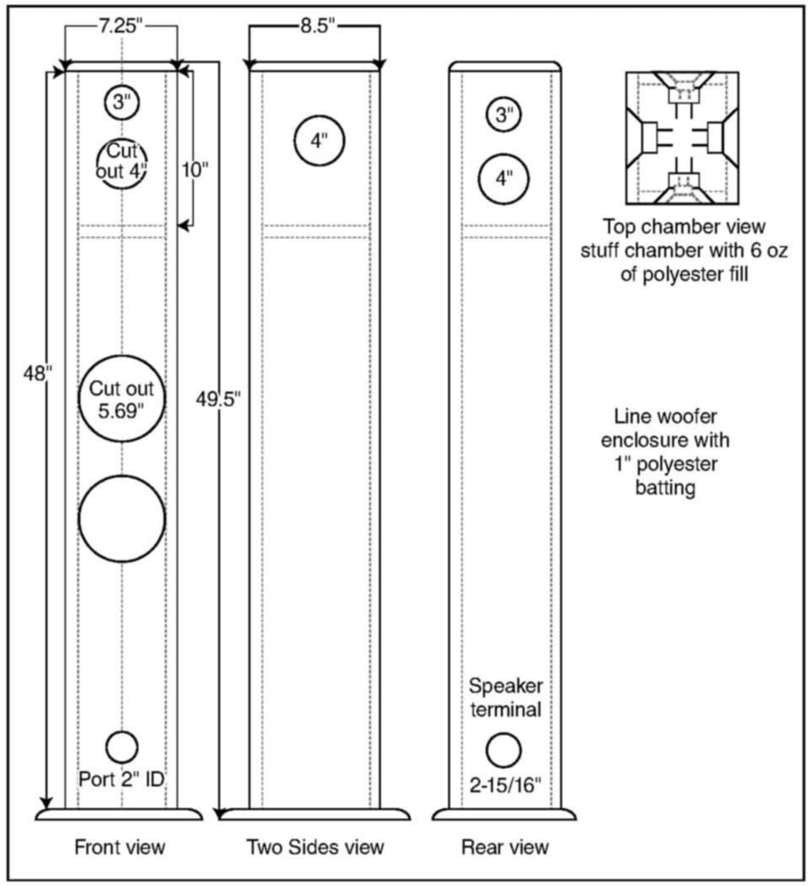


Figure 1: The drawing shows the dimensions of the enclosure.

Photo 4: Before making the driver cutouts, the speakers are laid out to determine the center points of the drivers. The woofer should be positioned as close as possible for good mutual coupling.

About the Author

Ken Bird has had a 40-year engineering and product management career focusing on consumer, industrial, and telecommunications electronics. Ken retired to his native Missouri and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (WOKLB), and digital photography.

with the Jasper Model 240, available from Parts Express (**Photo 5**).

Once I cut all the holes, I began assembly using corner clamps. Spread carpenters wood glue on both surfaces and then nail them together using 1-1/4″ nails (**Photo 6**). With the front and three sides together, I added the chamber divider and using a caulking gun caulk sealed all the joints. Then, I added the rear panel and caulked through the woofer holes and the still open top area.

I attached the top using both liquid nail caulk glue and 2″ finishing nails. Then, I attached the bottom with liquid nail caulk glue and 1-5/8″ drywall screws. Both the top and bottom were then hand caulked to assure a better seal.

I preinstalled the wiring and the crossover components along with the 1/2″ of polyester quilt batting, doubled up, and lining three sides of the woofer enclosure walls. I stuffed the sealed upper chamber with 5oz of polyester just prior to installing the last midrange driver.

Photo 5: A router with a 1/4″ bit with a Jason circle cutting jig was used to make the cutouts. The tweeters require a 3″ hole, the 4″ drivers require a 4″ hole and the woofers a 5.69″ hole. Cuts can also be made with a powered jig saw.

Photo 6: A nail gun with 2″ finishing nails and wood glue were used to fasten the panels. All internal panel seams were caulked. Make sure there are no air leaks between the upper and the lower sections of the enclosure.

The Crossover Circuit

The crossover circuit looks like a three-way design but it is a simple two-way, first-order Linkwitz crossing in the vicinity of 850Hz. The Piezo tweeters are in parallel with a filter to attenuate their turn on frequency. While the 4″ midrange speakers are advertised as having a response out to 15kHz, listening and measurements showed a reasonable flat response to about 8kHz, with a peak at 10kHz and a fast fall off to 15kHz. Using four of the speakers in series parallel configuration smoothed out the response and reduced the 10kHz peak. The Piezo tweeter with the filter adds brilliance to the upper range without the usual harshness associated with Piezo tweeters (**Figure 2**).

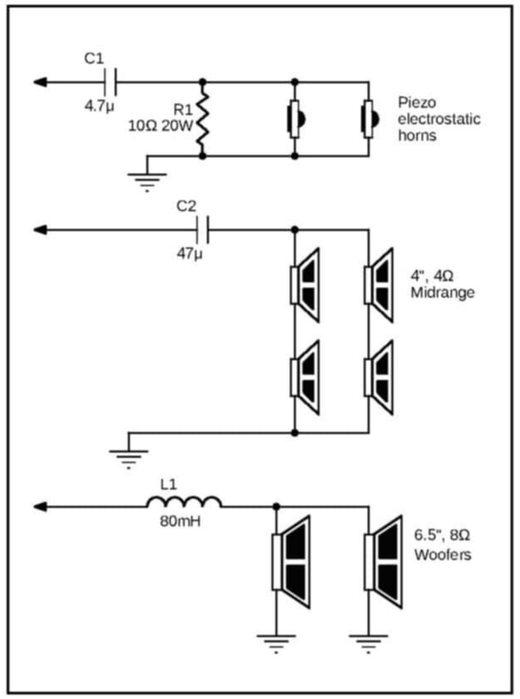


Figure 2: This is a schematic of the crossover components. The woofer coil was mounted in the lower chamber with the other components mounted in the upper chamber. This arrangement required only a single wire hole in floor of the upper chamber.

Finishing the Enclosure

The builder can choose any finish they desire, stain, paint, or even a wood veneer. I stained this system after sanding and used a wood sealer for the pine surface. This reduced the variations found in pine that will stain unevenly. The goal was to give the enclosures a distressed antique look. After staining, I applied a coat of polyethylene satin finish varnish (**Photo 7**).

Installing the Drivers

My enclosure was designed for a friend who has young children, who in the past had found it entertaining to poke holes in the drivers and flatten the center dust caps. Thus, each speaker is protected with a separate grille. The grille, 6.5″ frames, and the speakers are attached with the woofers nesting in the grille frames. I aligned the holes in the grille frames with those in the speaker and attached with 1″ drywall screws.

Before mounting, I lined each 6.5″ grille frame with a gasket made of 1/4″ foam door insulation.

Photo 7: I used 1/4" foam insulation tape to seal around the grille rims. The speakers have a built in gasket and do not need the foam. The 4" drivers also have gaskets but the Piezo tweeters will also need the foam applied.

Photo 8: A close-up view of the front and the side shows the grilles in place for the upper chamber drivers.

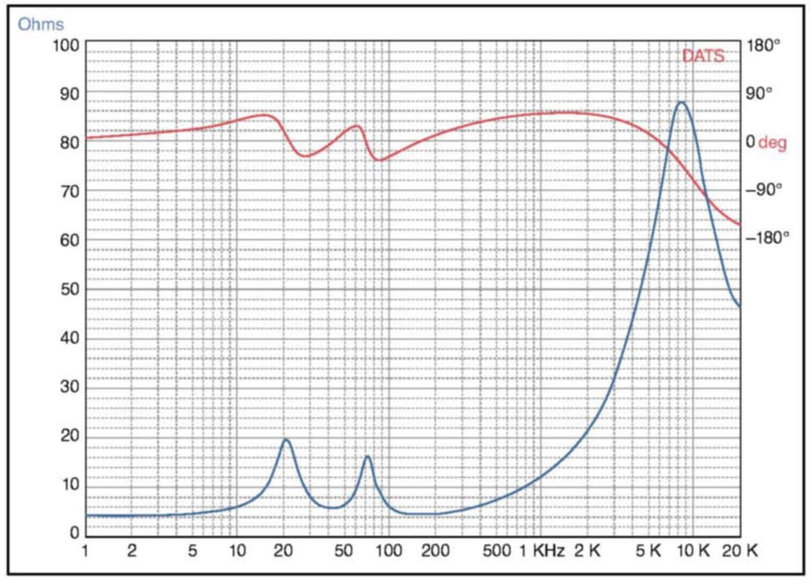


Figure 3: This graph shows the impedance of drivers in the enclosure.

The foam compresses, forming an air seal for the speaker. The grilles for the 4" speakers are actually designated 5", as the 4" model would not fit over pin-cushion-style speaker frame. No foam is needed on these grilles. Builders might eliminate the grilles and fashion their own grilles.

I found that the sound quality was not adversely affected by the coverings. There were no noticeable vibrations or audible diffractions affects due to the grilles (**Photo 8**).

Testing the Speakers

For the first test, I used an audio oscillator into an amplifier running from the low bass up through the high range to listen for any rattles or vibrations. Then, I connected the speakers to the receiver for listening tests. I didn't make any measurements except a tuning check using the Dayton system (**Figure 3**), which showed a 40Hz minimum between the two box impedance peaks.

The system had a very airy sound supported with a nicely balanced bass range. Imaging was good for vocalists with the system really spreading out orchestras and choruses with fine dynamics. It is hard not to like an omnidirectional speaker system.

Parts List

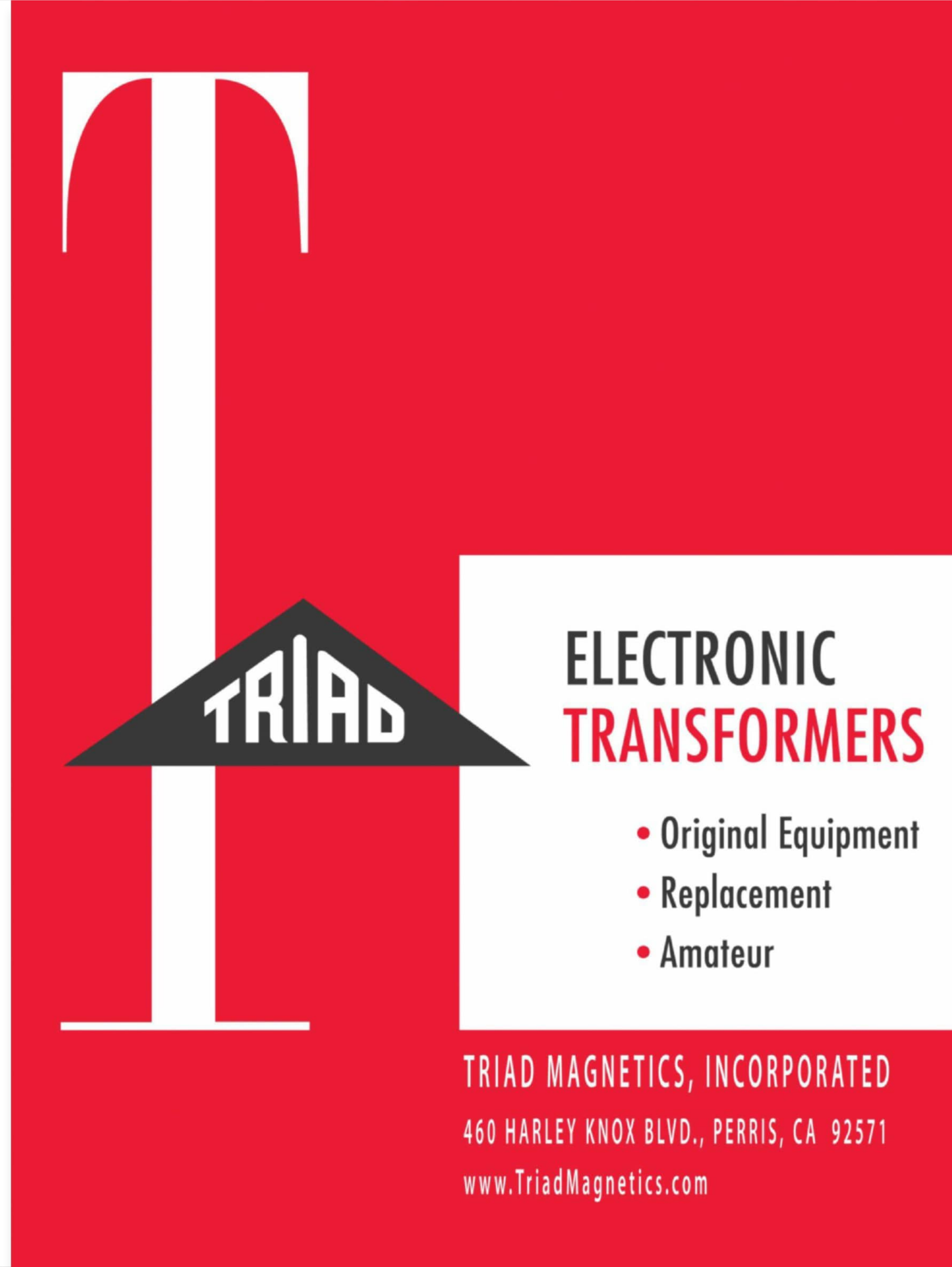
Enclosure Panels (for two enclosures)
Eight each 0.75" × 7.25" × 48 "panels
(Pine, plywood, hardwood, or MFD)
Two each chamber partitions 7.25" × 5.75"
(same material as the side panels)
Two each top 7.75" × 9.25"
Tow each base 11.25" × 13"

The Drivers
Four each Piezo Tweeters: GRS PZ 1005
Parts Express #292-442 (Hole cutout 3-1/16")
Eight each 4" Midrange: Dayton Model PC 105-4 4Ω
Parts Express #295-158
Four each 6.5" Woofers: Dayton Model 8Ω
Part Express #160-8 #295-305
4 each grille for 6.6" speakers: Part Express #260-446
8 each grilles for 4" speakers (Order from "Floratek"
on Amazon use 5" model)

Crossover Network
2 each 47µF non-polarized capacitors:
Parts Express #027-352
2 each coils 0.80mH: Parts Express #266-822
2 each speaker terminals Gold Posts:
Parts Express #260-283 (2-15/16" cutout)

Piezo Filter Parts
2 each non-polarized 4.7µF capacitors:
Parts Express #247-302

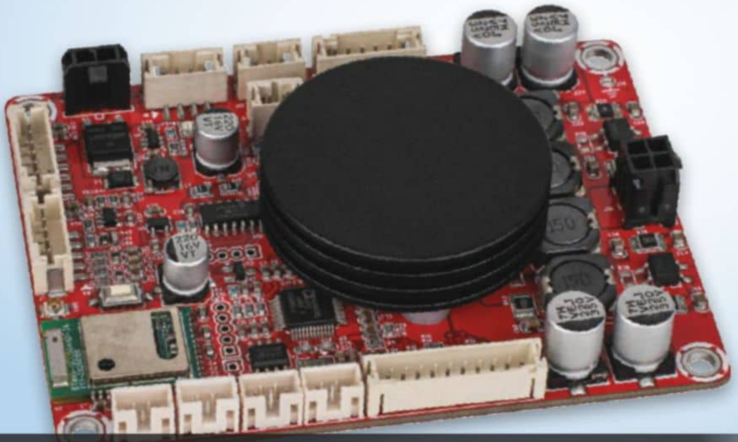
Lining and Stuffing
1 Bag Polyester pillow fill
1 Bag Polyester 1/2" Quilt backing



aptX HD Bluetooth Versatility

All In One... Digital Signal Processing

Make Your Own Sound



- ✔ Bluetooth 5.0 with AptX HD
- ✔ External Potentiometers
- ✔ DSP Processing
- ✔ Fully Customizable with SigmaStudio

Dayton Audio KABD-250 2 x 50W + DSP Amplifier Board aptX HD Bluetooth 5.0

The all-in-one combination of Bluetooth 5.0 aptX HD streaming, advanced signal processing, and 2 channels of efficient class D amplification all in a compact form factor, makes the KABD-250 the perfect choice for creating an incredible sound system that is truly your own.

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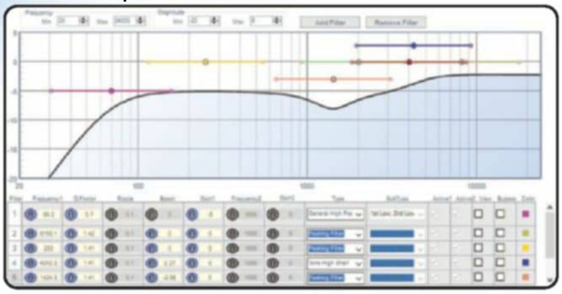
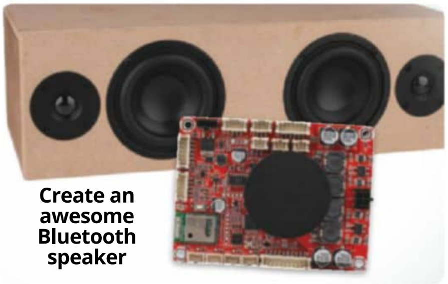
Dayton Audio DSPB-ICP1 In-Circuit Programmer for KABD-250 Boards

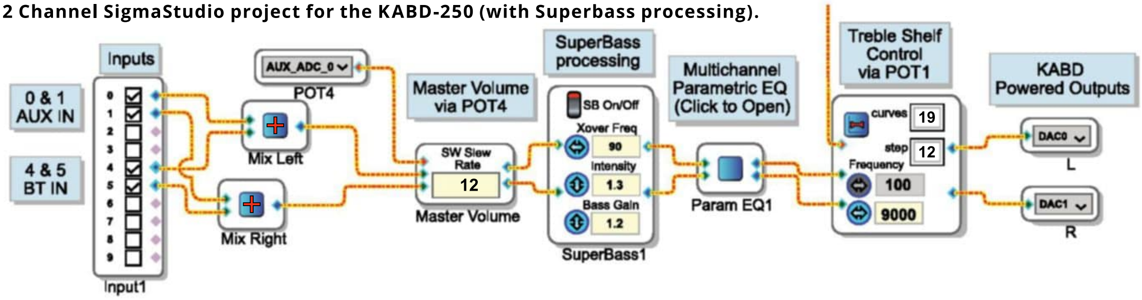
Parametric Equalizer





Create an awesome Bluetooth speaker

2 Channel SigmaStudio project for the KABD-250 (with Superbass processing).



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